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Call Setup Delay Analysis of H.323 and SIP

Master's Thesis in Computer Network Engineering

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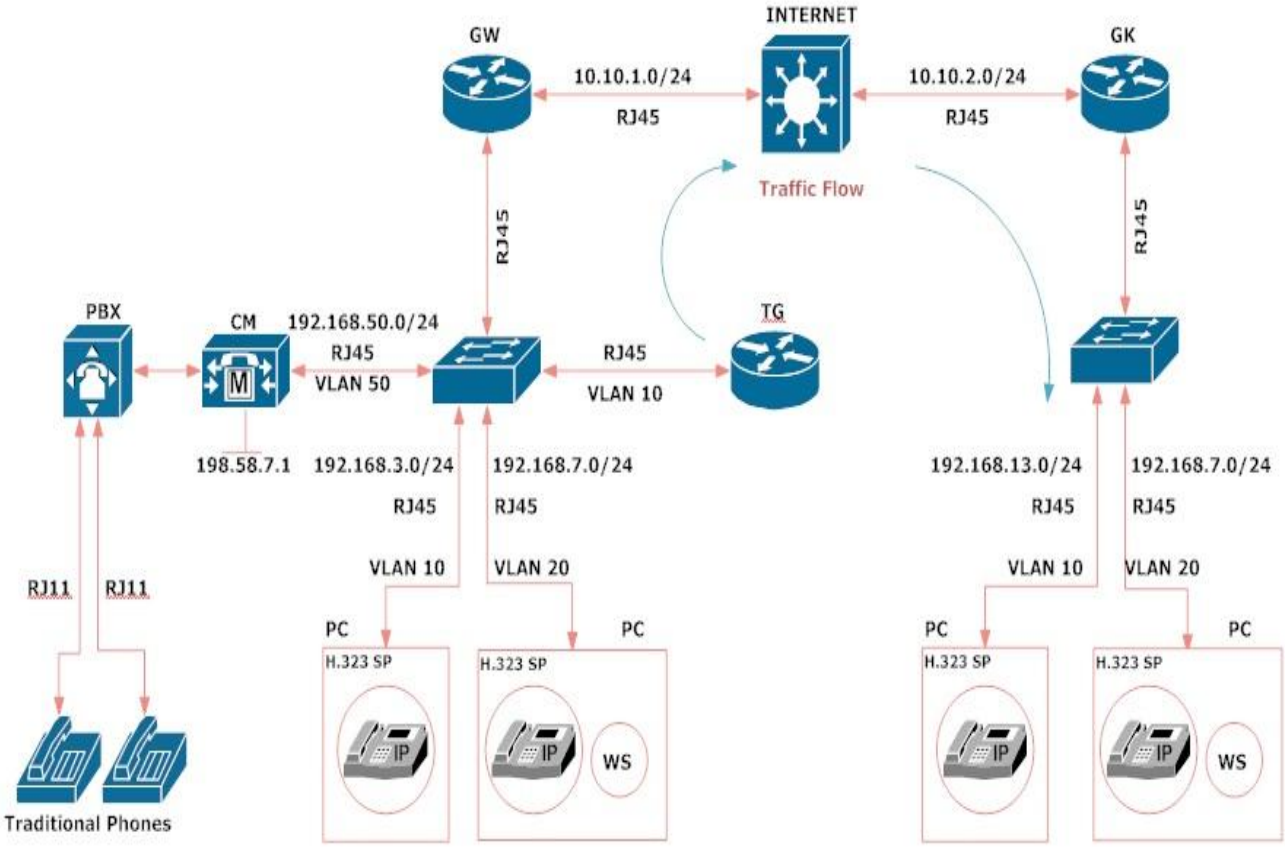

School of Information Science, Computer and Electrical Engineering
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Preface

We would like to thank Halmstad University for giving us a platform to complete our Master Thesis. We are also whole heartedly thankful to Tony Larsson and Wagner Ourique de Morais**,** who supported us in finishing our thesis work in a successful way.

We would also like to thank our fellow course mates who helped us in our studies.

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Abstract

IP Telephony is a technology which uses internet and signalling protocols like H.323 and SIP to setup and transfer voice signals from one destination to another. These protocols use various encoding schemes to transmit voice signals over the digital technology. The success of IP Telephony is the cost and time effectiveness; hence we studied on the call setup delay of these two signaling protocols H.323 and SIP.

H.323 uses TCP and SIP uses UDP as transport protocols to set up a call. They have different scopes that distinguish them. The signalling protocols H.323 and SIP call setup delays are observed and studied successfully between the traditional telephones and IP telephones. The call setup delay for H.323 is analysed experimentally in the Cisco environment by establishing calls at various loads such as null load, medium load and heavy load. Due to practical difficulties the SIP call setup delay is studied only literally. The parameters that we considered for our experiments are the average call setup delay, bandwidth and different traffic loads. Finally, the results of both H.323 and SIP are compared with each other to find out the best signalling protocol. The comparison is done according to the call setup delay and other factors like complexity, compatibility, reliability and bandwidth utilization.

.

List of Acronyms

ATM	*Asynchronous Transfer Mode*
ACELP	*Algebraic Code Excitation*
CO	*Central Office*
CM	*Cisco Call Manager*
DUAL	*Diffusing Update Algorithm*
E1	*E-Carrier*
EKTS	*Electronic Key Telephone System*
EIGRP	*Enhanced Interior Gateway Routing Protocol*
FTP	*File Transfer Protocol*
GW	*Gateway*
GK	*Gatekeeper*
HDLC	*High Level Data Link*
HTTP	*Hypertext Transfer Protocol*
IP	*Internet Protocol*
ITU	*International Telecommunication Union*
IETF	*Internet Engineering Task Force*
IEEE	*Institute of Electrical and Electronics Engineers*
IGRP	*Interior Gateway Routing Protocol*
KTS	*Key Telephone System*
LAN	*Local Area Network*
LLC	*Logical Link Control*
LS	*Location Server*
MTP	*Message Transfer Part*
MGCP	*Media Gateway Control Protocol*
MAC	*Media Access Control*
MCU	*Multipoint Conference Unit*
MC	*Multipoint Controller*
MP-MLQ	*Multipulse Maximum Likelihood Quantization*
OSI	*Open System Interconnection*
PSTN	*Public Switched Telephone Network*
PBX	*Private Branch Exchange*
PPP	*Point to Point Protocol*
PS	*Proxy Server*

QOS	*Quality of Service*
RTP	*Real-time Transport Protocol*
RAS	*Registration, Admission and Status*
RTCP	*Real-Time Transmission Control Protocol*
SIP	*Session Initiation Protocol*
SDP	*Session Description Protocol*
SS7	*Signaling System 7*
SMTP	*Simple Mail Transfer Protocol*
SP	*Soft Phone*
T1	*T- carrier*
TG	*Traffic Generator*
TCP	*Transmission Control Protocol*
UDP	*User Datagram Protocol*
UAC	*User Agent Client*
UAS	*User Agent Server*
VoIP	*Voice over Internet Protocol*
VLAN	*Virtual Local Area Network*
WAN	*Wide Area Network*
WS	*Wireshark Tool*

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1 INTRODUCTION

1.1 Application Area and Motivation

The technology recently introduced in the telecommunication industry is IP telephony. The major concept of IP telephony is to transfer voice messages through networks using internet protocol. The main task of IP telephony is to transmit the multimedia messages such as video and voice in the form of data packets. IP telephony follows the recommendations created by the Internet Engineering Task Force (IETF) and International Telecommunication Union (ITU). The major benefit of internet telephony is to reduce the charges of long distance calls to the local call rate. In addition to this, it also transmits all kinds of data types like video, voice and data in a single medium by assuming the IP protocol as the mutual protocol. [1][2]

To support voice over internet telephony protocols such as H.323, SIP, MGCP, Megaco are used along with the number of encoding techniques (CODECs) like G.711, G.723, G.726, G.728 and G.729 according to the user's requirements and the capability of the network. Even though IP Telephony is cost effective, it has some factors that affects it better qualities such as packet loss, call setup delay, packet delay, Jitter and end to end delay.

In IP telephony, the first factor which affects the quality of service in transmitting voice packets is the call setup delay. The call setup delay is the time interval between the last digit dialed and receiving the ring tone back. The call setup delay depends on the transport protocol used for call establishment. [11]

H.323 and SIP are the most competing signaling protocols of IP telephony in which H.323 works on TCP and SIP works on UDP. The two major signaling protocols H.323 and SIP use different transport protocols TCP and UDP for transmitting voice packets over the internet. This specific concept motivated us to work on the call setup delay time properties of the signaling protocols H.323 and SIP over IP telephony.

1.2 Problem Statement

In telecommunication, IP telephony is successful technology invented to transfer voice packets over the internet and also between IP telephones and traditional telephones. It is successful because of cost and time effectiveness. To handle IP telephony more efficiently, we have many signaling protocols; and the most competing protocols are H.323 by ITU and SIP by IETF. [7]

H.323 and SIP work on different conditions and have different features. For example, H.323 uses TCP for connection establishment whereas SIP uses UDP. They also have their own merits and demerits. As we mentioned earlier in this section, IP Telephony is good because of it features such as low cost and less call setup delay. However, if the organisations looking for the services of H.323, which is very expensive and complex then why are they not ready to use SIP?

1.3 Goals and Expected Results

The major goal of our thesis is to find out the call setup delay on both signaling protocols H.323 and SIP.

A sub goal is to setup a test environment between local exchange and foreign exchange using H.323 signaling protocol in the Cisco environment. Then we make a number of calls between

the two exchanges over three types of traffic to find out the average call setup delay of H.323 signaling protocol.

The call setup delay for the SIP signalling protocol is studied theoretically. Finally the experimental results of H.323 call setup delay is compared with the theoretical call setup delay of SIP signalling protocol that thus only is studied theoretically.

The expected result of our thesis is that the signalling protocol H.323 may consume more time than the SIP signalling protocol measured call setup delay. However the call setup delay of two signaling protocols might also be very similar.

2 BACKGROUND

2.1 IP Telephony Background

In a basic telephone system, a call can be made by a user only with the help of basic telephone. If a user wants to make a call to the destination, the call should pass various stages before it reach the destination. The steps are the call should reach the company internal phone system or to the Public Switched Telephone Network (PSTN) through analog trunk or through digital trunk like T1/E1. After this the call reaches the destination from PSTN. Here the calls are transmitted through transmitting equipment's like copper cables, fibre optic cables, microwave communications and satellite communications. [2]

The important traditional telephony equipment's are a KTS- Key Telephone System, EKTS-Electronic Key Telephone Systems, PBX- Private Branch Exchange, CO- Central Office and Toll Switch which helps to implement the traditional telephony system. [2]

Later this telephony system was advanced to the new IP telephony method in which the internet plays a major role with the help of signalling protocols like H.323, SIP, and so on and also with various encoder schemes derived by the ITU and the IETF. This telephony system reduced the cost and effort significantly when compared to the basic telephony system. [2]

2.2 Signalling System 7 (SS7)

SS7 is an ITU-T standard came to use in 1987. SS7 is for the administration purpose of the traditional telecommunication systems. Message Transfer Part (MTP) and Signalling Connection Control Part (SCCP) are the two major parts of SS7 and because of SCCP errors and frauds are reduced and also the call setup and take downs are performed faster as it works out of band. [2]

The two major functionalities of SS7 are that it allows the end party know the details of the called party and also it connects the call setup fast with the help of dedicated circuit switched connections. [2]

2.3 Voice over IP Protocol

2.3.1 Voice over IP

Voice over Internet Protocol (VoIP) significant benefits as it uses Internet Protocol for the transport mechanism to make calls from a regular telephone or from soft phones. VoIP transmits packets between the same service providers and also it allows the making of calls to a normal telephone number. The data transmission over the internet between various devices is performed successfully by the concept of an OSI model. [4]

2.3.2 The OSI Open System Interconnection

- Physical Layer

The physical link between the called party and the end party is activated, maintained and deactivated by the physical layer which sets electrical, mechanical, procedural and functional specifications. The categories in physical layer implementations are LAN and WAN specifications. [2]

- Data Link layer

The reliable service of transmitting the data through the physical link layer is carried out by the data link layer. Network and protocol characteristics such as physical addressing, network topology, error notification, frame sequencing and flow control are varied according to the specifications mentioned in the data link layer.

The two sub layers present in the data link layer are,

1. MAC (Media access control)
2. LLC (Logical link control)

The LLC sub layer which supports both connectionless and connection oriented network is defined by the IEEE 802.2 specifications. The function of LLC is to manage the connection between the devices. The functionality of MAC sub layer is to manage the protocol access to the physical network. In order to identify one device among the multiple devices present in the network, it enables the MAC address which is unique to each device, defined by the IEEE MAC specifications. [2]

- Network Layer

The network layer implementations such as Internet Protocol (IP), defines the network address which completely differs from the MAC address. As the logical network layout is defined by this layer, the routers can easily determine the way to forward the packets. Hence at the layer 3, all the design and implementation works for the internetworks are implemented. [2]

- Transport layer

In the transport layer, the data is received from the session layer, they are then analysed for errors and then the data is segmented and aligned in a sequence before they are dropped in the transport layer to transmit the data through the network.

This layer performs a major functionality called flow control which manages the data transmission between the devices according to the sender and receivers adoptability. Other than this function, multiplexing and error checking are also performed by this transport layer. TCP and UDP are the two protocols used in the internet as transport protocol. [2]

- Session Layer

Establishment, management and termination of sessions between the end users in the network is done by the session layer using service requests and service responses. The protocols implemented at the session layer manage the requests and responses co-ordination according to their functionality. Some examples are ZIP, SCP, Appletalk protocol and DECnet phase IV session layer protocol. [2]

- Presentation layer

The presentation layer handles the functions like coding and conversion that are applicable for the data from the application layer. These two functions check whether the data sent from one application layer is readable by the other application layer present at the end user.

Common data representation formats, conversion of character representation formats, common data compression schemes and common data encryption schemes are the coding and conversion schemes available in the presentation layer. [2]

- Application layer

The application layer is the one in which the user directly interacts with the application layer through software applications which implement a communicating component. Telnet, File Transfer Protocol (FTP) and Simple Mail Transfer Protocol (SMTP) are some of the implementations complete in the application layer. [2]

2.3.3 Voice traffic Management by OSI

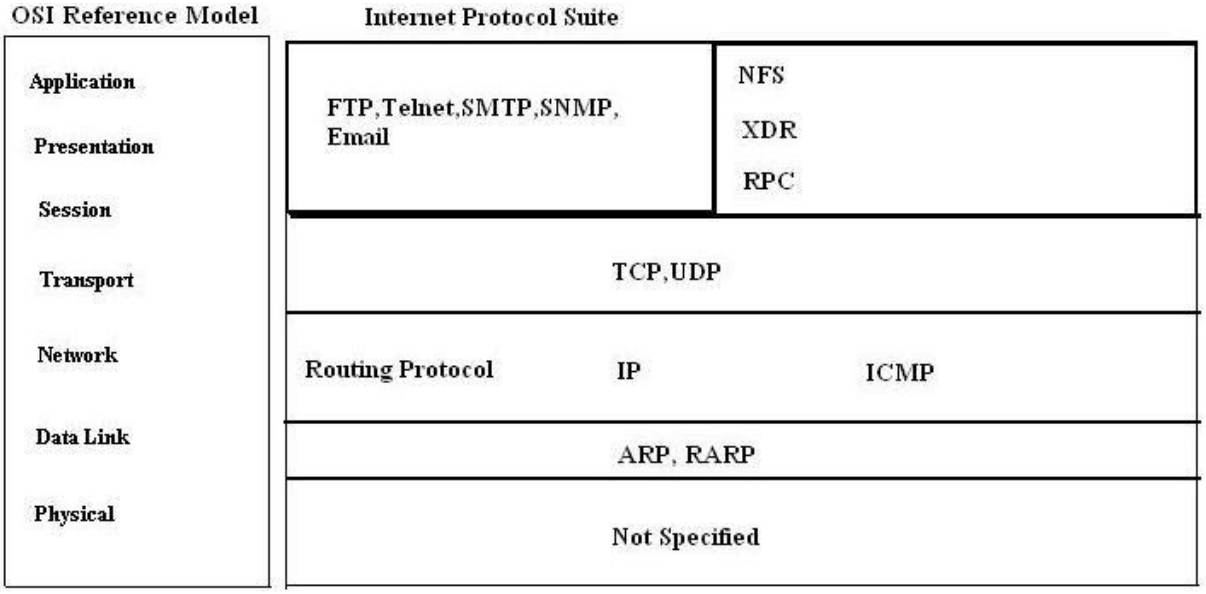

Fig (2.3.3) Voice Traffic management by OSI [2]

In the Application layer, applications such as CISCO IP Communicator and Call Manager which provide interface to users to generate voice at the PCs and then they convert and compress the voice signal before passing it to the network. Human speech is considered as an application and a standard telephone is considered as a user if a gateway is used. [2] [8]

In the Presentation layer, the CODECs are implemented. The CODECs are used to compress the voice. There are lots of encoding techniques from which the user can select and negotiate the CODEC which they required according to their process. [2] [8]

In the Session layer the process of implementing the signaling protocol is being handled. The end to end call signaling methods are defined by H.323 and SIP. The process of separating the signaling function from the voice call function is defined by the protocols MGCP and Megaco/H.248 and this process is said to be a client/server model for voice signaling. Call agents are used to control signaling instead of end devices in client/server architecture while the central control device manages only the call setup function. [2] [8]

OSI Layer	**Protocols and VoIP Components**
Application	IP Communicator, Call Manager and Human speech
Presentation	CODECs
Session	H.323, SIP,MGCP and Megaco
Transport	RTP and UDP (media), TCP and UDP (signal)
Network	IP
Data link	Frame Relay, ATM, Ethernet, Point to Point Protocol (PPP), Multilink PPP, T1, E1, ISDN BRI, ISDN PRI and High Level Data Link (HDLC) which supports transport IP packets

Physical	Category 5 shielded twisted pair (UTP), Coaxial cable, Ethernet crossover cable and RJ11 are some physical technologies from which the suitable technology that supports the transport of data link frames can be chosen here.

Tab (2.3.3) OSI layer and VoIP components [2] [8]

To carry the voice traffic across the network, VoIP implementation has a standard method of using RTP inside UDP. When the packets reach the destination they are unsynchronized and out of order. Once the packets reach the end user it should be resynchronized and reordered before playing. The services like sequence numbers and time stamps are not provided by the UDP, hence RTP plays a role along with UDP as it provides these functionalities. [2] [8]

The voice packets are ready to transmit across the IP network when they are encapsulated at the transport layer. Using any kind of data link layer and physical layer which are capable of transmitting the data, the IP traffic from the network layer is transmitted across the network to the destination. [2] [8]

2.4 H.323

H.323 is a standard recommended by the International Telecommunication Union (ITU) and was developed in May of 1996.It works to transmit voice, video, data and fax communication across an IP-based network connectivity maintained with the PSTN. It encourages the compatibility in videoconference transmission over IP networks. H.323 was initially promoted as a way to provide a better performance than other signalling protocols in audio, video and data packet transmission in the event when LAN did not provide QOS. [7]

H.323 has four components such as give below

- Terminals: Probably either traditional telephone or soft phones.
- Multipoint conference unit: One of the major duties of MCU is the conference management. Two major components of the MCU are the Multipoint Controller (MC) and Multipoint Processors (MP) which helps in managing the multipoint conference. The MC does not perform the multiplexing of audio and video but with the help of H.245 it traces the capabilities of the end users. As well as this, the MP handles multiplexing of data streams with the help of the MC. [7]
- Gateways: This translates the traffic to the format which is required to pass the packets over the internet at the time of using traditional phone to make a call without the trouble of the type of traffic. It is the actual endpoint of the network which makes the two way communication between the end users in the IP network. [7]
- Gatekeeper: It performs various processes like centralized call management; call admission control, management of bandwidth, address translation for source and destination, authentication and user location because it is the important component of H.323 which acts like a manager within its specific zone. [7]

2.4.1 H.255

The tasks such as registration, admission and status (RAS) are performed by the H.225 standard. To ensure the availability of the connections between the endpoints, gateways and gatekeepers, the RAS protocol is used. The RAS is used only when the gatekeeper is available in the network. The RAS handles the tasks such as registration, admission control,

bandwidth changes, and status and withdraw procedures between endpoints and gatekeepers. [2]

2.4.2 Q.931

Q.931 participates in call control and so it is used to establish a connection between the terminals and also frames data, because it is a link layer protocol. The major task of the protocol is to define how each H.323 layer should interact with the peer layers, hence according to the agreed formats, the participants can exchange information. Q.931 is a part of H.225. From larger channel, Q.931 can define a logical channel by using a specific method. With the help of protocol discriminator, Q.931 can easily identify the messages with its call reference value and message types. The methods of receiving and processing the Q.931 messages are specified by H.225 layers. [2]

2.4.3 H.245 Control Signaling

To connect the H.323 compatible terminals, the H.245 control signalling is used to provide the call control mechanism. As it carries the control messages to govern the operations over H.323 endpoints, the H.245 channel is a reliable channel. The information that are carried by the control messages are

- Capabilities Exchange
- To open and close logical channels those are used to carry media streams
- Preference requests
- Flow control messages
- General commands and indications [2]

2.5 SIP

SIP is an application control protocol designed by IETF to establish VoIP connections, which can create, modify and terminate a connection within the end users. The process of SIP is also similar to HTTP architecture which manages the connection among the end users with the requests and responses. Simply to say, it is Client-Server architecture. The server and client exchange the requests and responses to establish a connection. The SIP has two messages called INVITE and ACK to open a reliable connection through which the call data are passed. [7]

To carry out the negotiation for codec identification SIP depends on SDP (Session Description Protocol). To allow the users to work on a set of compatible media types, SIP supports session descriptions. By the process of proxy and redirecting the requests to the user's current location SIP also helps in User Mobility. [7]

The services provided by SIP are,

User Location: The end system should be determined for communication.

Call Setup: The establishment of call, by ringing from the calling party to the called party.

User Availability: It determines whether the called party is interested to communicate or not.

User Capabilities: To determine the kind of media and the required parameters to support that media are carried over here.

Call handling: To transfer and to terminate the calls are handled here.

2.5.1 Why SIP

While comparing SIP with H.323, the SIP protocol is simple because it never requires full compatibility but for H.323 it requires full compatibility. H.323 handles the messages with binary representation whereas SIP uses only the normal textual representation. SIP is highly very standard and highly scalable but H.323 is not as much as SIP. In SIP loop detection is simple with H.323 and also the header space is only 37 in SIP when compared with H.323 which uses more than hundred. Hence many telecom industries looking trying to use SIP in the place of H.323. [7]

2.5.2 SIP Components

1) User Agents
2) Network Server

- User Agents

On behalf of a User, the user agent acts as an end System. It has 2 parts, Client and Server. Client is known as User Agent Client (UAC) and Server as User Agent Server (UAS). UAC indicates the SIP Request and the UAS receives the SIP request and sends responses to the UAC on behalf of user.

- Network Server

Here we use 3 servers together in a network.

Registration server: Registers the current location of the user

Proxy Server: It receives requests and forwards them to the next hop server, which provides more information about the called party.

Redirect Server: This server on receiving the request determines the next hop server and returns the address of the next hop server to the client instead of forwarding the request. [7]

2.5.3 Overview of SIP operation

The messages are used for the communication between the client and the server. Those messages are,

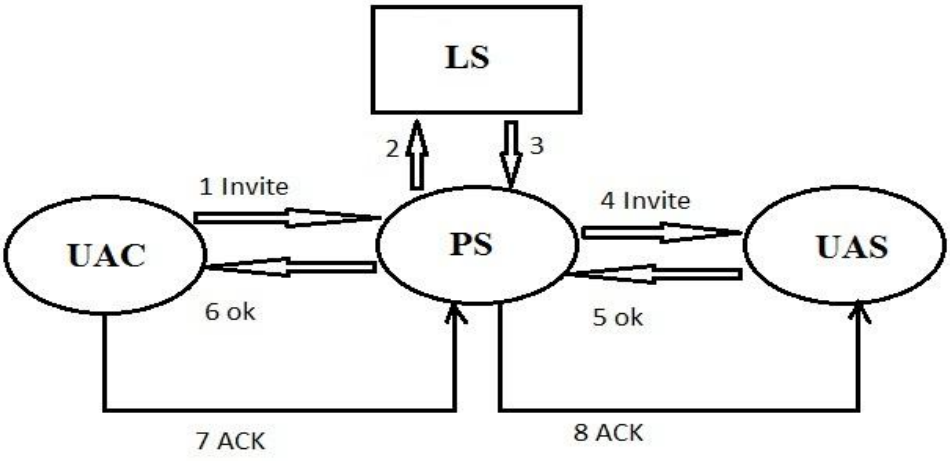

Fig (2.5.3) Formation of SIP operation [7]

SIP Messages

- INVITE: invites a user to make a call
- BYE: terminate a connection between the 2 end users
- ACK: makes a reliable exchange of invitation messages
- OPTIONS: to find information about what capabilities do the call has.
- REGISTER: the SIP registration server gets the details about the location of the user.
- CANCEL: it terminates the process of searching for a user.

The SIP address is used to identify the callers and callees. The caller first identifies the SIP server and sends a request. The caller can connect the callees directly or through some servers called redirect servers. The calls are uniquely identified by the call ID field in the SIP message. [7]

- SIP Addressing
  The SIP URL which is in the form sip:username@host is used to identify the SIP host. The SIP address can be for an individual or for a group.
- Locating the SIP server
  The request from the client can be sent directly to the IP address or port corresponding to the Uniform Request Identifier UDP or to the SIP proxy server. [7]
- SIP Transaction
  The client can send requests to the server when the host part of the Request URI has been resolved to a SIP server. Then a SIP transaction can be made triggering a request along with the response for the request made by the client. Through a reliable TCP or through an unreliable UDP the can be sent between the end users. [7]
- SIP Invitation
  An INVITE followed by ACK is the two requests required to perform a successful SIP Invitation. To make a callee participate in a specific conference call or in a two party conversation, the INVITE request is used. By sending an ACK request from the caller the caller confirms the callee, after the callee agreed to participate in the call. A session description in the INVITE request provides enough information to the called party to join the session. By sending a similar description to the invitation as a response, the callee expresses its wish to join the call. [7]
- Locating a user
  According to the time, the callee may keep on change its position. The SIP server is used to register the locations of the callee dynamically. A list of locations is sent to the caller when it sends any queries regarding the callee's location to the SIP server. To perform this process a server called Location server is used in the SIP system to generate and pass the list to the SIP server. [7]
- Changing an Existing session
  By re-issuing the INVITE message with same Call ID but a new body to convey the new information is sent if there may any need to change the parameters of an existing session. [7]

2.6 Comparison of H.323 and SIP

Features	H.323	SIP
Extensibility	The versions of H.323 are extended by the ITU standard with various new features and in such a way not to affect the existing features. H.323 V1, H.323 V2 and H.323 V3 are developed according to the requirements of the user and are done in such a way to preserve backward compatibility.	IETF standard holds the in charge in creating new features to SIP, by not affecting the existing features. Even though SIP is extended by its new features it grows within itself by not having any various versions to support the backward compatibility.
Reliability	To eradicate and to reduce the network failure, H.323 handles various methods like alternate gatekeepers and alternate endpoints, which are backups for the active gatekeepers and endpoints.	In SIP, if the network failure occurs due to proxy server, it can be analyzed only through the time expiration by the user agents. Because SIP do not have any failure handling procedures. Hence the user agent sends request to the other proxy server which may leads more time delay in connection establishment.
Administrative Requirements	A call can be established directly among the endpoints even without the help of a gatekeeper. Gatekeepers are used only for registration and address resolution gatekeepers are used.	The duty of SIP proxy is to make registration, address resolution and call routing. So the SIP can establish a call without the help of SIP proxy among the user agents.
Message Encoding	Here messages are encoded in a compact binary format which suits narrowband and broadband connections. With the help of widely available decoders the messages are effectively encoded and decoded.	ASCII text format is used in encoding SIP messages which is readable by human. These encoded messages are large which is not suitable for the network where bandwidth, delay and processing are a concern.
Media Transport	RTP/RTCP, SRTP	RTP/RTCP, SRTP
Scalability	Through one RAS message exchange the address resolution is provided when gatekeeper is used in H.323 or it can also route all call signaling traffic. The direct call model method is used to connect the end points	To perform the address resolution for a SIP device using a SIP proxy, at least 3 complete messages have to be exchanged and handled by the proxy for every call. The number of data on wire may be more on large

	directly in large networks.	networks like IMS networks.
Call Forking	The gatekeeper in H.323 can control call signaling and also fork the call to many devices simultaneously.	In SIP proxies can control the call signals and fork them to a number of devices.
Multicast Signaling	Yes it is performed by sending location request (LRQ) and auto gatekeeper discovery (GRQ).	Yes it is done through group INVITEs.
Minimum VoIP call Ports	3 (call signaling, RTP, RTCP)	3 (SIP, RTP, RTCP)
Media Topology	Unicast, Multicast, Star and Centralized	Unicast, Multicast, Star and Centralized.

Tab (2.6) Comparison of H.323 and SIP [5]

2.7 EIGRP

Enhanced Interior Gateway Routing Protocol is a Cisco routing protocol, which can only be used in a Cisco network. As it has easy configuration, strong reliability and fast access it has overcome its drawbacks. EIGRP is similar to RIP which uses Distance Vector algorithms to determine the path. EIGRP uses some additional metrics that are not available in RIP which makes EIGRP more special. EIGRP can handle any sized network as its metric is based on the bandwidth and net delay on each possible bandwidth. To reduce the loops that occur, EIGRP uses another algorithm called Diffusing Update algorithm (DUAL). [4]

The major task of DUAL is to ensure the routing tables of all routers to be free from loops, and also to ease the local sharing among equal cost links. For this EIGRP allows the routers to make use of several different possible paths, that have the same metric. This DUAL algorithm differentiates EIGRP's features from its old version of Cisco protocol IGRP. [4]

EIGRP can recover quickly, if any changes on the network topology have occurred, because of the regular updates to the routing tables of each router. EIGRP works very efficiently over large networks, hence it consumes very little bandwidth because EIGRP distributes information to the router only when the changes happens in the topology. In the remaining times it sends only "Hello" packets between the routers to check whether the routers are alive or not in the topology. [4]

3 SOLUTION TO BE INVESTIGATED

In this section we going to describe the factor to be tested and investigated in IP telephony. Our thesis is to investigate the call setup time of H.323 and SIP signaling protocols.

3.1 Call Setup Time

The signaling performance has been improved rapidly in the 120 years history of telephony. In 1928, Gerardi and Jowett declared the call setup time as 1.2 minutes, which was reduced from 4 minutes declared in 1923. In 1978, Duffy and Mercer reported that the average time between end of dialing and ring back was 10.9 seconds. In 1998, it was further reduced to less than 2 seconds for toll calls and for calls which requires database lookups it was claimed as 2.5 seconds by AT&T. [11]

But in the case of IP Telephony which was high speed backbone links must provide less call setup delay compared to the SS7 system which was 64 kbps links. [11]

3.1.1 Call setup time

- Dial to Ring Delay

Call setup time is the time interval between entering the last dialed digit and receiving ring back. Compared to traditional telephony with IP Telephony in call setup, IP Telephony has an advantage which traditional telephony does not have. With-traditional telephony, there is no acoustic feedback between dialing and ringing. This makes the caller aware the error in the call when it takes more time than average to connect the call. [13]

According to the ITU 721 standard, an average delay for local, toll and international calls should not be more than 3.0, 5.0 or 8.0 seconds respectively. [13]

The type of call that the user needs to establish differentiate the importance of call setup delay. [13]

- Post Pick Up Delay

Another important delay which plays a major role in call setup is the post pick up delay. Post pick delay is defined as the interval between when the destination users pick up the receiver and the source user receives the indication of the destination pick up. According to ITU 721 recommendations the average post pick up delay should be 0.753, 1.53 and 2.0 seconds for local, toll and international connections. But 95% valuations are as 1.5, 3.0 and 5.0 seconds. [13]

Call setup is the number of round trips essential to communicate the audio between the caller and the callee. In H.323 v2 it was reduced to a decent, acceptable delay due to a fast call setup procedure. A relatively small delay in call setup is found in version second of H.323 and SIP, and hence the two provides a very different call setup process. [13]

3.1.2 Call Setup on H.323 and SIP

H.225/Q931 signaling procedures are used by H.323 to establish a call connection between the caller and the callee H.323 v1 calls take about 6 to 7 round trips in the call setup process, which includes the setting up H.323 TOP connection and Q931, and this depends according to the usage of a gatekeeper. [12]

H.323 v2 the fast call setup method, the call setup delay is reduced up to 3 round trips which includes the process of establishing a call and initial media streams and it is achieved by the performance of the H.323 logical channel information in the SETUP and CONNECT

manager. Voice communication based on G.711 codec only can be handled between the calling parties with this fast call setup method, because they cannot exchange the capabilities. 11.245 optionally perform the capability exchange procedure only when the calling parties would like to use other media channel types and it can also be done after the establishment of G.711 channels. [12]

Like SIP H.323 v3 also works on UDP to transfer call setup messages. H323 v3 can also work on TCP whereas SIP cannot. The important advantage of UDP is, in establishing a transport layer connection no round trip delay occurs. In UDP the call setup delay is around 1.5 to 2.5 round trips and again this depends on the involvement of the gatekeeper. [12]

The performance of SIP call setup is similar to that of H.323 v3 but H.323 v3 has an advantage over SIP. If UDP fails H.323 can perform the call setup establishment through TCP, which performs simultaneously along with UDP. When the call setup using UDP is successful, it automatically stops the performance of TCP effectively. However, SIP runs TCP and UDP setups sequentially. If UDP fails, it takes some time to establish the call setup with the help of TCP; hence because of this process, there is an increase in call setup delay in SIP compared to H.323 v3. [12]

3.2 H.323 Call Setup

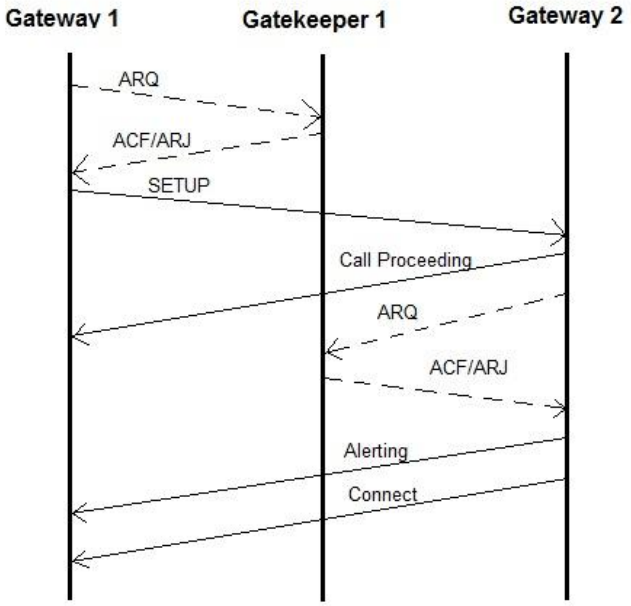

Fig (3.2) call setup of H.323 [3]

- - - - RAS Messages

_____Call signaling messages

- The caller (Gateway 1) exchanges the admission request (ARQ) and admission confirmation (ACF) with the Gatekeeper.
- The call signaling address of callee (Gateway 2) will be then returned by the gatekeeper with ACF to the gateway 1.

- Using the transport address the gateway 1 then send the call setup messages to the gateway 2.
- After this the call will be established and start to proceed the call.
- The gatekeeper will exchange an ARQ/ACF message with the gatekeeper if it wishes to accept the call.
- For this request the gatekeeper will send an ACF/ARJ response message.
- Then the gateway 1 receives an alert message from gateway 2. Instead of ACF message if the gateway 2 receives admission reject (ARJ) message from the gatekeeper, the gateway 1 receives the release complete message from gateway 2 instead of getting the alerting message.
- Finally the gateway 2 makes a response with the connect message to the gateway 1. [3]

3.3 SIP Session Setup

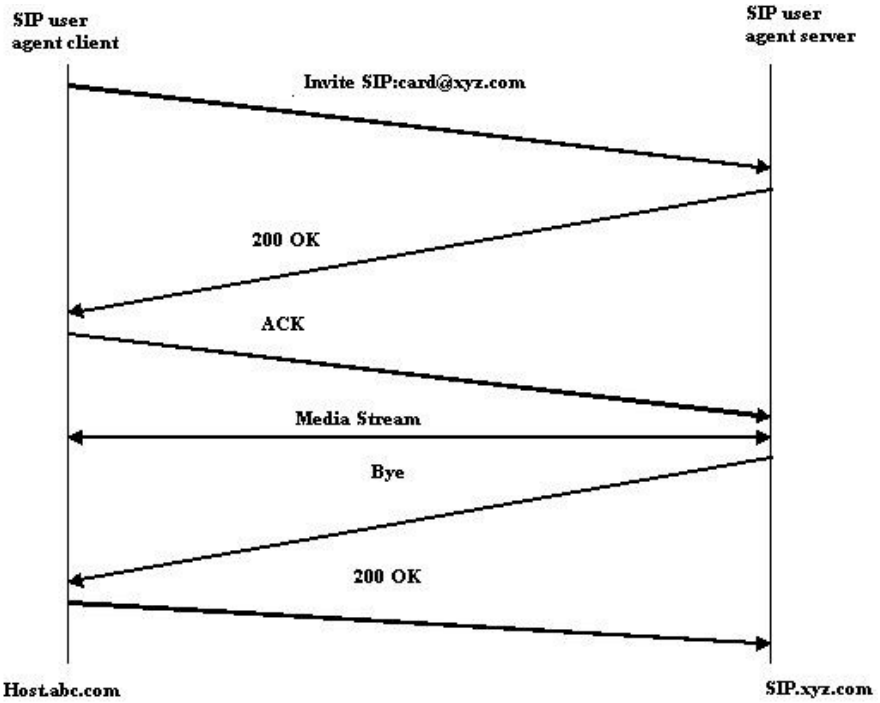

Fig (3.3) Call setup of SIP without Proxy [9]

This call flow diagram represents the call setup between the client and server using SIP without a proxy server. This type of call setup resembles a HTTP connection establishment with no representatives to handle the connection between them. [9]

The diagram shown below represents the call setup made by SIP signalling protocol using a proxy server. Here the call setup is handled by the proxy server as a representative between the client and server. This proxy server takes care about the establishment and termination of the call on the network between client and server. [9]

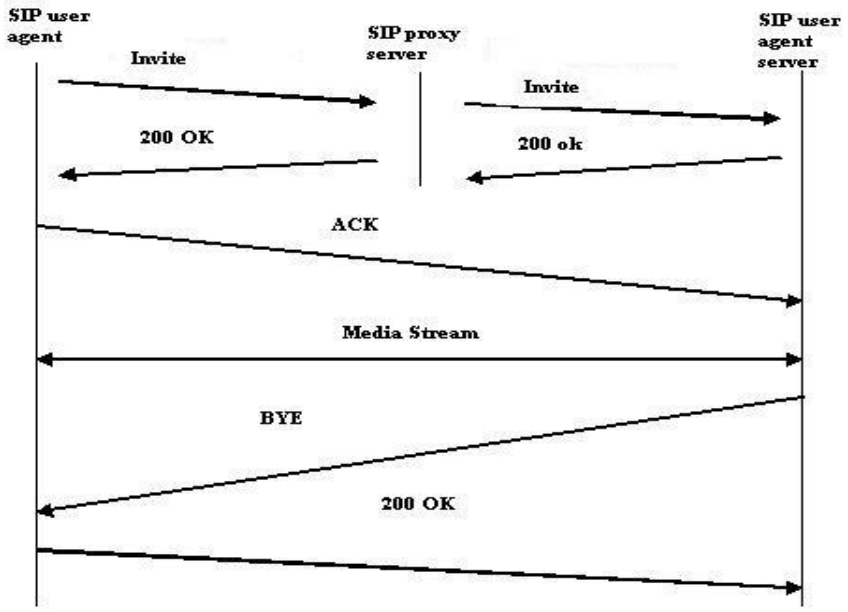

Fig (3.3) Call setup of SIP with Proxy server. [9]

3.4 G.723.1

The encoding scheme G.723.1 is a very low bit rate codec recommended by ITU-T standard. It is officially named as Dual rate speech coder for multimedia communications transmitting at 5.3 and 6.3 kbits/sec. It is an audio codec which can encode and compresses the voice audio signals to 30ms frames. It also has an algorithmic look ahead of 7.5ms which results in over all algorithmic delay of 37.5ms. The other delays due to this codec are the time spent on processing the voice data in encoder and decoder, the time taken to transmit the data over the communication link and the extra buffering delay taken for multiplexing protocol. [10]

This codec is optimized to present the high quality voice data at the mentioned rate with limited amount of complexity. This codec works on two algorithms either Multipulse Maximum Likelihood Quantization (MP-MLQ) or Algebraic Code Excitation (ACELP). When it uses MP-MLQ it consumes 6.3 Kbits/sec (24 byte frames) and 5.3 Kbits/sec when it uses ACELP (20 byte frames). [10]

4 NETWORK TOPOLOGY

In this section we going to describe the network topologies used to test the H.323 signaling protocol and used to study the SIP signaling protocol. We also going to describe the process performed in each topology.

4.1 Network Topology of H.323 Call Setup

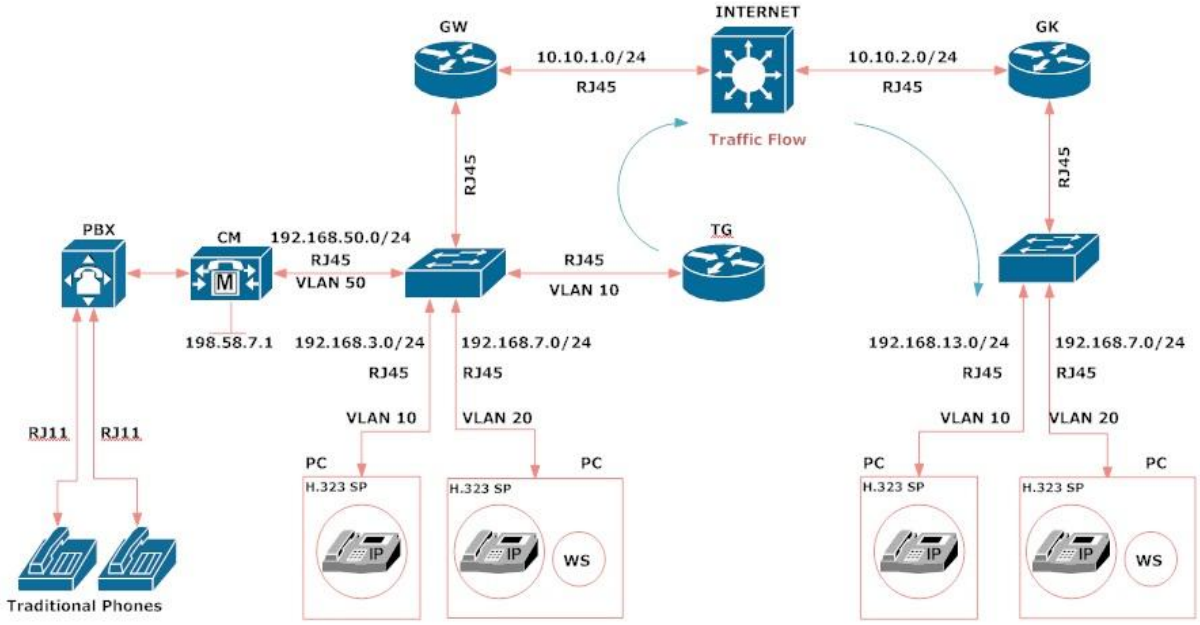

Fig (4.1a) Real time topology for H.323

The figure (4.1a) shown above is the real time network topology built to analyze the call setup for the signaling protocol H.323 in the Cisco lab. The devices we used to construct this topology are H.323 soft phones, Cisco 2800 series routers, Cisco Pagent routers, Cisco switches, PBX, traditional phones RJ11 and RJ45 cables to establish the connection.

The H.323 soft phones represent the end users which are connected to the switches over Vlan 10 and Vlan 20. The PC connected at Vlan 10 handles the data and the PC connected at Vlan 20 handles the voice. Their IP addresses are 192.168.3.0/24 and 192.168.7.0/24 respectively. Then, to the same switch a router which is configured as Call Manager is connected over the Vlan 50 with the IP address 192.168.50.0/24 and also another router which is configured as the gateway and is connected over the Vlan 80. Then that router Gateway is connected to the router Gatekeeper through the Internet cloud. The IP address for the Gateway and Gatekeeper are 10.10.1.0/24 and 10.10.2.0/24 respectively. The destination soft phones are then connected to the gatekeeper. The whole network capability is connected with 100 BaseT Ethernet cables. EIGRP routing protocol is used to communicate with the whole network.

Here in this topology, we used a PBX which handles the traditional phones and this PBX is connected to the Call Manager which is connected to the internet via the gateway. Now all the 3 destinations are connected to each other. Calls can be generated from any source to destinations.

To estimate the call setup time according to the real time process we built a traffic generator in our topology and it is connected to the switch over Vlan 10 which can generate the traffic

from the PC which supports data. To simulate the call setup time at various conditions, traffic is generated at different loads. One is at no load and the second one is at load 200 (medium) and the third one is load 1000 (heavy). The bandwidth allotted to transmit all the packets from traffic generator through this network topology is 32 Kbps.

To estimate the traffic and to calculate the call setup time, a free and open source network analyzing software called Wireshark is used at both the local and foreign exchanges, because it captures each and every actions performed in the topology like the REQ and ACK messages passed to connect the call, packets transferred, the time taken to setup the connection and the protocols used.

The Cisco IOS software for Gatekeeper and Traffic generator are installed in Cisco pagent routers. The IOS for call manager and gateway are installed in Cisco 2800 series routers. Wireshark is software which is installed in the workstation connected to the gateway and gatekeeper.

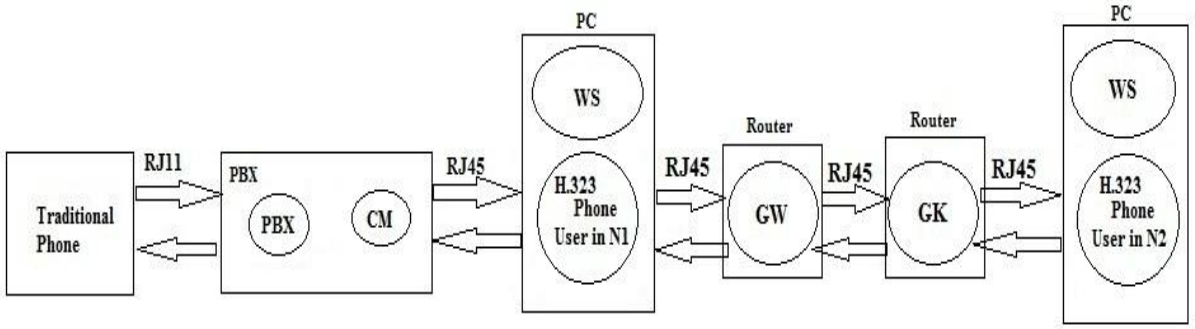

Fig (4.1b) Logical structure of H.323 topology

= Hardware device

= Software installed on Hardware device

This logical diagram defines the process performed in the H.323 network topology. We experimented our H.323 test environment according to the logical diagram shown above. We fixed the phone numbers as 5556001 and 5559 for traditional phones connected to PBX. Then the soft IP phone number fixed to the gateway is 5551 and the soft IP phone number connected to the gatekeeper is 5552. To find out the call setup time of H.323 signaling protocol we made calls between the traditional phones and the soft IP phones. In this H.323 test environment, RJ11 cable is used to connect the traditional phone to the PBX and RJ45 cables are used to connect the other devices in the network.

The calls made from each user passes through the gateway and gatekeeper. When the user in N1 calls to the traditional phone, it takes messages to gatekeeper through gateway. Then it communicates to the call manager (CM) to pass the voice packets to the traditional phone connected to the PBX.

4.1.1 Call Flow Diagram of H.323

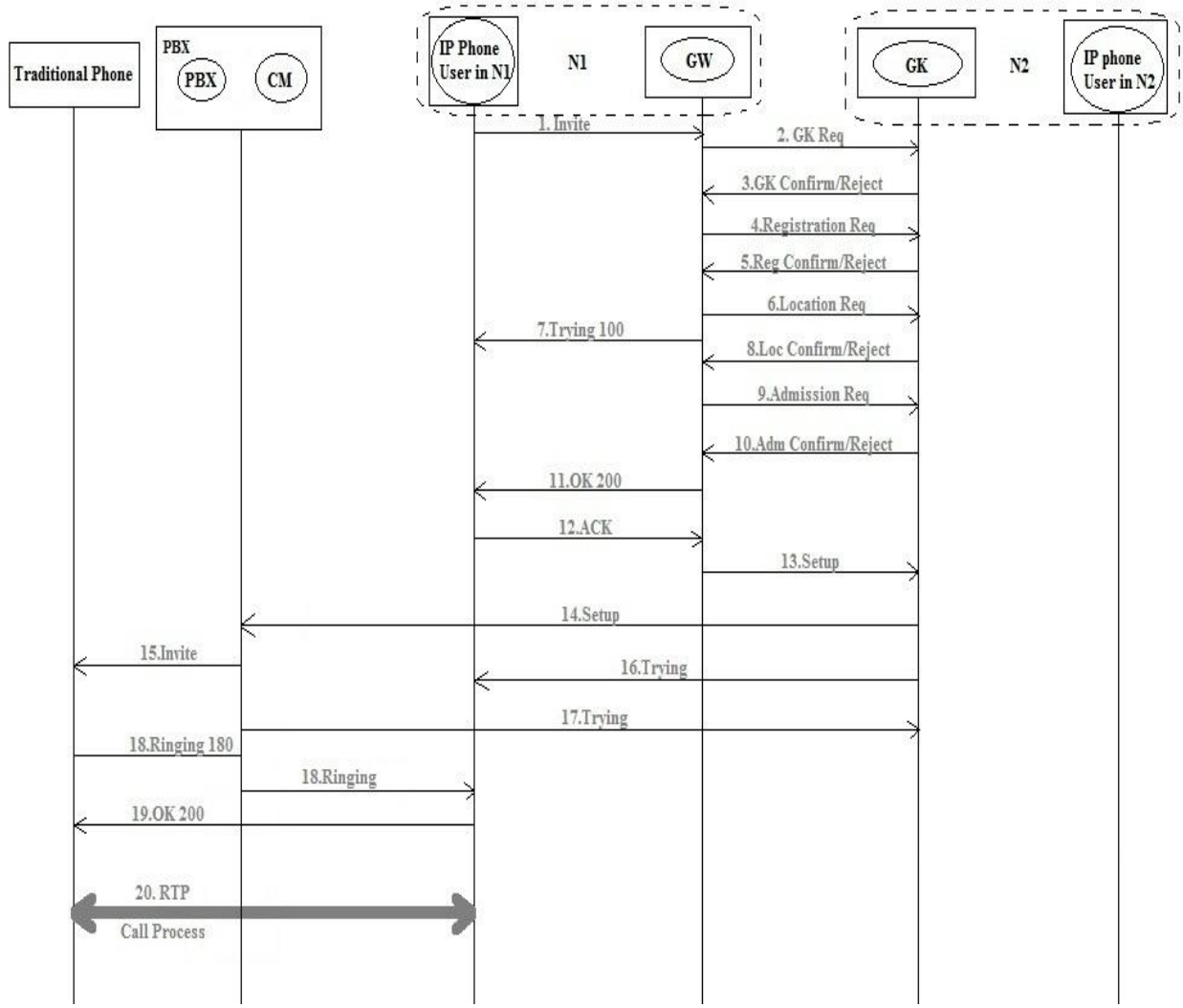

Fig (4.1.1a) Call flow diagram between User A and Traditional phone

The fig (4.1.1a) shown above represents the call flow processes performed during the call setup between the user A and traditional phone. When the user A wants to communicate with the traditional phone it send and INVITE message to the gateway. The communications between gateway and gatekeeper mentioned in the call flow diagram are the REQ and ACK for confirming the registration, location and admission control to the user A. when the admission is confirmed, the gatekeeper starts to communicate with the CM to connect to the traditional phone. Here the traditional phone is connected to the PBX with RJ11 cable. When the traditional phone accepts the invitation from the user A, it starts to communicate by transferring voice packets between them through RTP channel. The connection establishment process is done by the TCP protocol by sending REQs and ACKs between the terminals.

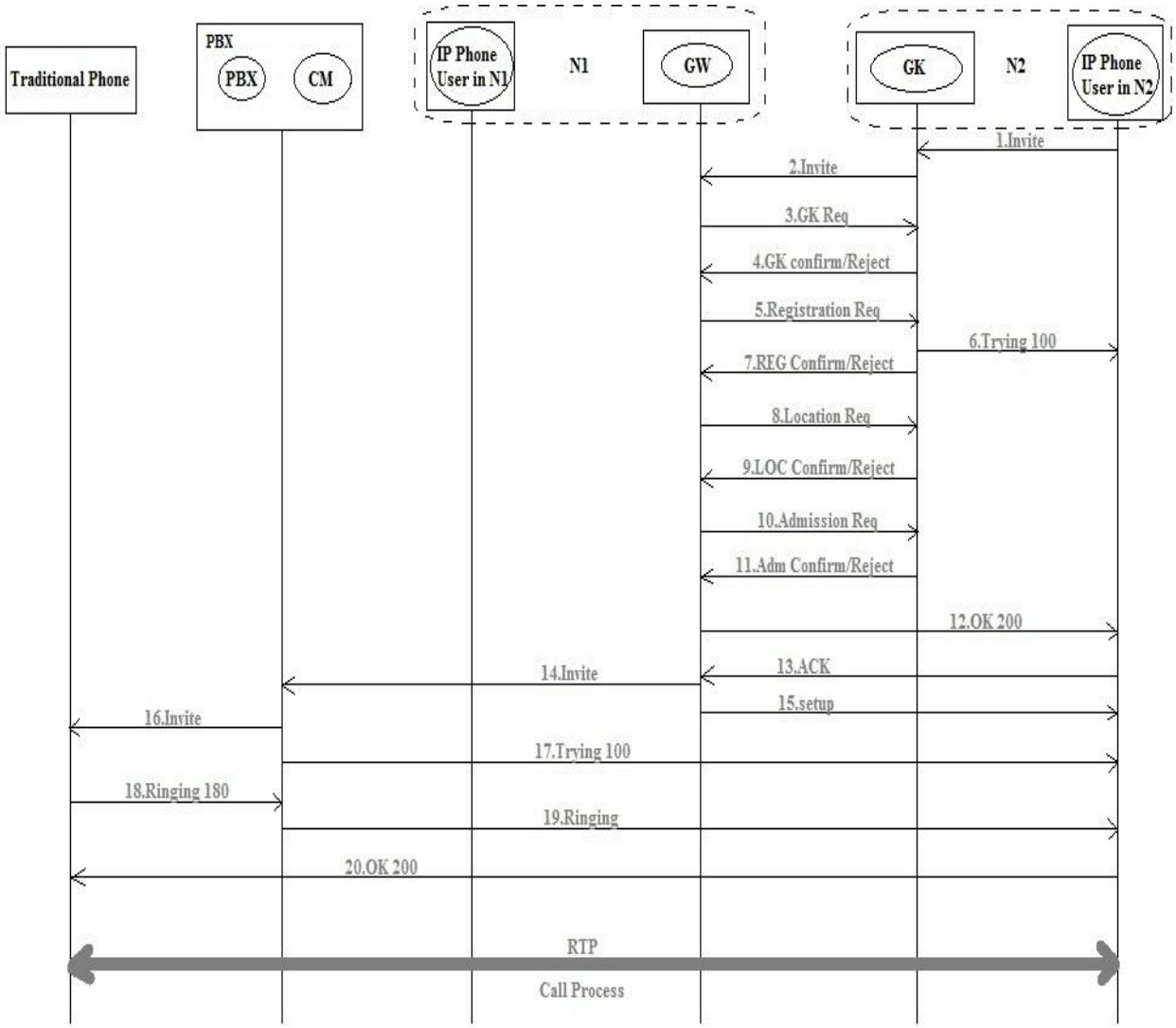

Fig (4.1.1b) Call flow diagram between User B and Traditional phone

The fig (4.1.1b) shown above represents the call flow between the user B and the traditional phone. The connection establishment process performed between user B and traditional phone are same as like the process done between user A and traditional phone explained in the section 4.1.1a.

4.2 Network Topology of SIP call setup

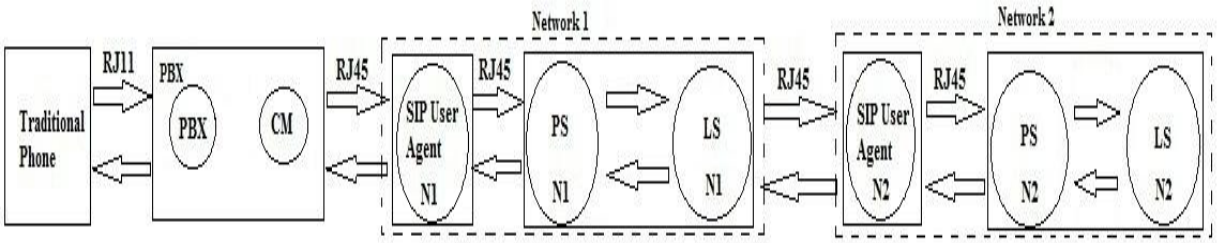

Fig (4.2) Logical Network topology of SIP

The logical topology shown in the fig 4.2 represents the network which works on a SIP signaling protocol. This topology is similar to the logical H.323 topology shown in the fig 4.1b but the differences are the use of a SIP proxy server and some other technical operations. In the SIP structure a SIP proxy server is used in each network domain instead of H.323 gatekeeper. H.323 uses H.245 as the call controlling protocol which controls the session whereas SIP uses SDP to handle the sessions. The most important is perhaps transport layer protocol, because SIP uses UDP where as H.323 uses TCP.

In SIP topology when a user needs to make a call from one network to the user in another network it has to pass through the proxy server. The proxy server finds the path of the end user with the help of the inbuilt location server. This may takes some time in connecting the call according to the distance and the number of proxy servers between the both users.

4.2.1 Call Flow Diagram of SIP

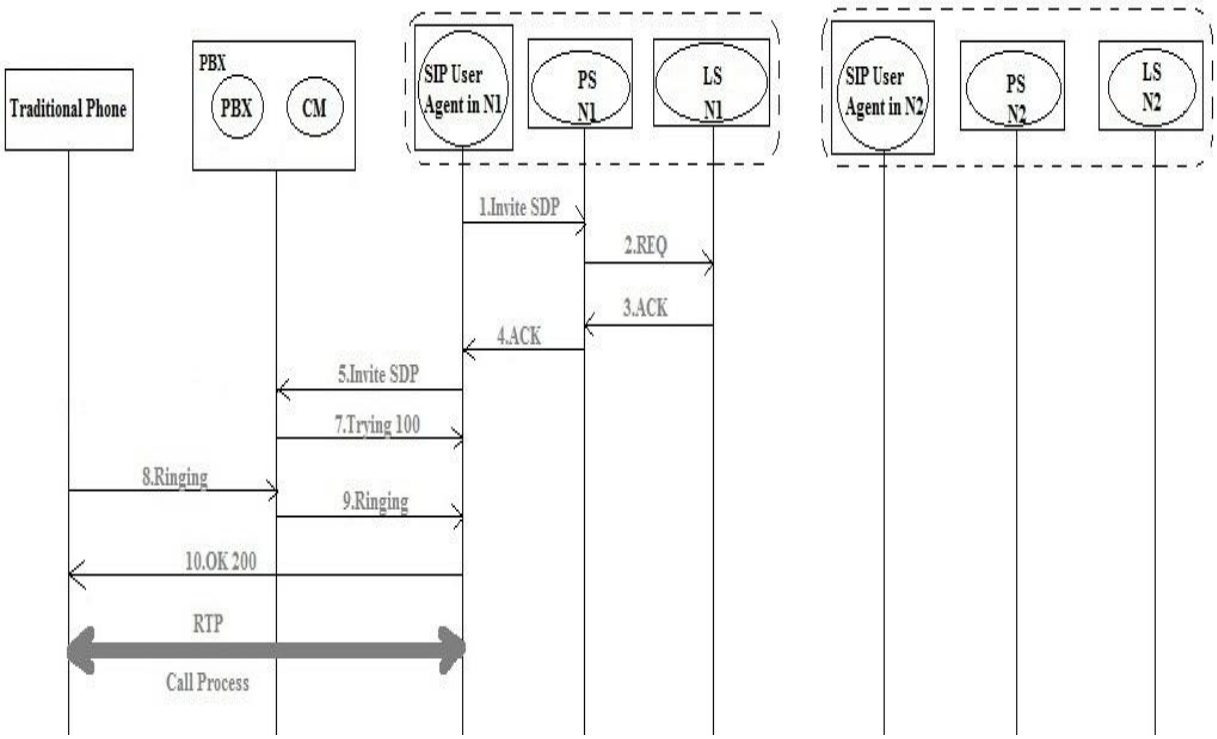

Fig (4.2.1a) Call flow diagram between SIP user agent A and Traditional phone

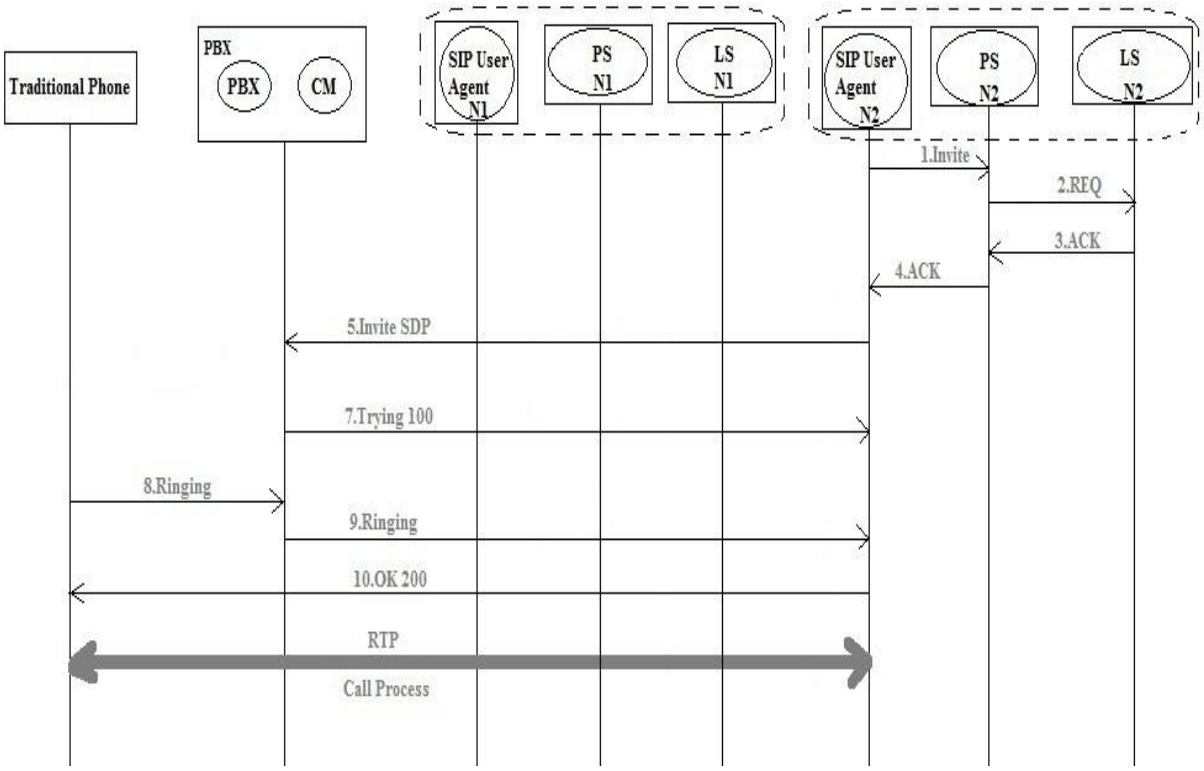

Fig (4.2.1b) Call flow diagram between SIP user agent B and traditional phone

The fig 4.2.1a and 4.2.1b shown above represent the connection establishment between the SIP user agent A and traditional phone; and between the SIP user agent B and traditional phone. The call connection process is done with the help of UDP protocol. When the user agent A wants to communicate with the traditional phone, it passes the message through proxy server. The proxy server gets the location of the destination from the location server which is inbuilt with the proxy server. Every network domain has its own proxy server. When the traditional phone accepts the invitation from the SIP user agent A, they start to communicate with each other through the RTP channel.

Features in H.323 and SIP Topology

Features	**H.323**	**SIP**
Gatekeeper	Yes	No
Proxy Server	No	Yes
UDP/IP	Yes	Yes
Session Description H.245	Yes	No
Session Description SDP	No	Yes
EIGRP	Yes	Yes

Features in H.323 and SIP Topology

The table (4.2.1) shown above represents the features that are possible and available in both the signaling protocols that are taken for analysis. The descriptions for each feature are described in the previous section 3.

4.2.2 Session Description

- H.245

  For the session description, H.245 is used in H.323, where it can manage media capabilities, establish and tear down media channels and also provide conference floor control. A set of capability descriptors, listed in decreasing order of preference is used to describe media capabilities in H.245. A capability descriptor is also known as simultaneous capability set, which is a set of alternative capability sets. A list of algorithms is available in each capability sets, from which one can be used at any given time. These lists of algorithms help the terminals to support audio, video and data simultaneously. [8]

- SDP

  SIP uses SDP for session description process. It holds a list of media types and the supported encoding schemes for each. The cross media and inter media are not expressed by SDP, like in H.245. SDP also cannot express that in conjunction with certain video codecs, certain audio codecs to be used. Most of the media capabilities in SIP can be described by H.245 but it is hard for SIP to describe the capabilities handled by H.245. Using the 'multipart' content type many SDP messages can be carried in the message body of SIP INVITE requests and responses. [8]

4.3 Transmission Control Protocol (TCP)

Transmission control protocol (TCP) was developed in 1984 and it is defined by RFC-793. TCP is the most mature protocol, which is considered to be more familiar in use across the internet world for data transmission. TCP is a connection oriented protocol which sends REQ and receives ACK between the end users to establish a connection. The connection with TCP is more reliable because of the functionality in arranging the data packets to be transmitted in the same fixed sizes and providing a sequence number to each packet before transmitting them over the internet. Because of this sequence number, the data packets are sent in a particular to the destination. [2]

If any data packet is lost while transmitting, immediately the destination will send a retransmission request to the source to resend that particular lost data. The destination can easily identify the lost data with the help of the sequence number attached to each packet. Hence because of this process TCP provides a reliable form of data transmission which is essential amongst internet users. [2]

In VoIP technology, TCP can play its role only up to the call setup process between the end users. Once the call is established, it automatically adopts the RTP protocol transmission running over UDP which can handle voice packets, because TCP cannot transfer voice packets along with the data packets. The other major features of TCP are error detection, congestion control, flow control and reliability. [2]

4.4 User Datagram Protocol (UDP)

UDP is a transport layer protocol defined to use with the IP network layer protocol. It is defined by RFC 768. To the end system, UDP provides a best effort datagram service. It is an unreliable service, it provides no guarantees for data delivery and the duplication is not protected. UDP and the UDP-Lite variant are similar when compared with other protocols in connecting the end systems. The end to end connection between communicating end systems are not established before by them. [2]

UDP is popular because of tunnelling protocol, where the packets of another protocol are encapsulated inside UDP datagram from one tunnelling endpoint and then they are

transmitted to another tunnelling endpoint these UDP datagrams are then expanded and forwarded as original packets. The tunnelling service in UDP helps in establishing virtual links that directly connect to distant locations in the physical network. UDP provides no communication security over the network. To create protection for the packets from eavesdropping, tampering, or message forgery additional protocol mechanism which provides security services should be configured in the network. [2]

4.5 REAL-TIME TRANSPORT PROTOCOL (RTP)

Real-Time Transport Protocol (RTP) is developed by IETF and established in 1996.It provides the transport of real-time data packets. RTP allows the protocol to easily adapt new versions of audio and video.

RTP allows end-to-end transport services in order to transmit the real-time data. These services include the features given below

1. payload type identification
2. sequence numbers
3. Time stamps

RTP and RTCP both can be used together in conjunction. RTP carries media such as video and audio, whereas RTCP is used to monitor transmission statistics and also provide (QOS) information. RTP is used for transfer of multimedia data on the other hand RTCP control information and QOS parameters. [6]

The major drawback of UDP is when the receiver receives the packets from sender they are mostly unsynchronized and unordered. Because UDP does not provide the services like sequence number and time stamps. Hence to synchronize and reorder the packets in UDP the RTP service is very much important. These protocols are configured to our topology for our expected results and analysis. [2]

5 RESULTS

5.1 Call Setup Results on H.323

In the Cisco lab, with the help of Cisco routers and switches, the real time network topology as shown in fig 4.1 was built to analyse the call setup using H.323 signaling protocol. The signals which passed through this network are encoded by G.711 encoding scheme. The run time analyses chart created by Wireshark network analyser shown in fig 5.1 represents the successful connectivity of the network. From this chart, before the data is being transferred, the devices and the protocols in the network are announcing themselves that they are active to transmit the signals across the network. Here the messages like 'Hello' and 'Keep Alive' of EIGRP, TCP, Skinny and SCCP are passed between the endpoints to mention that they are active. So the fig 5.1 shows that the network topology in fig 4.1 is successfully built and it is in running state.

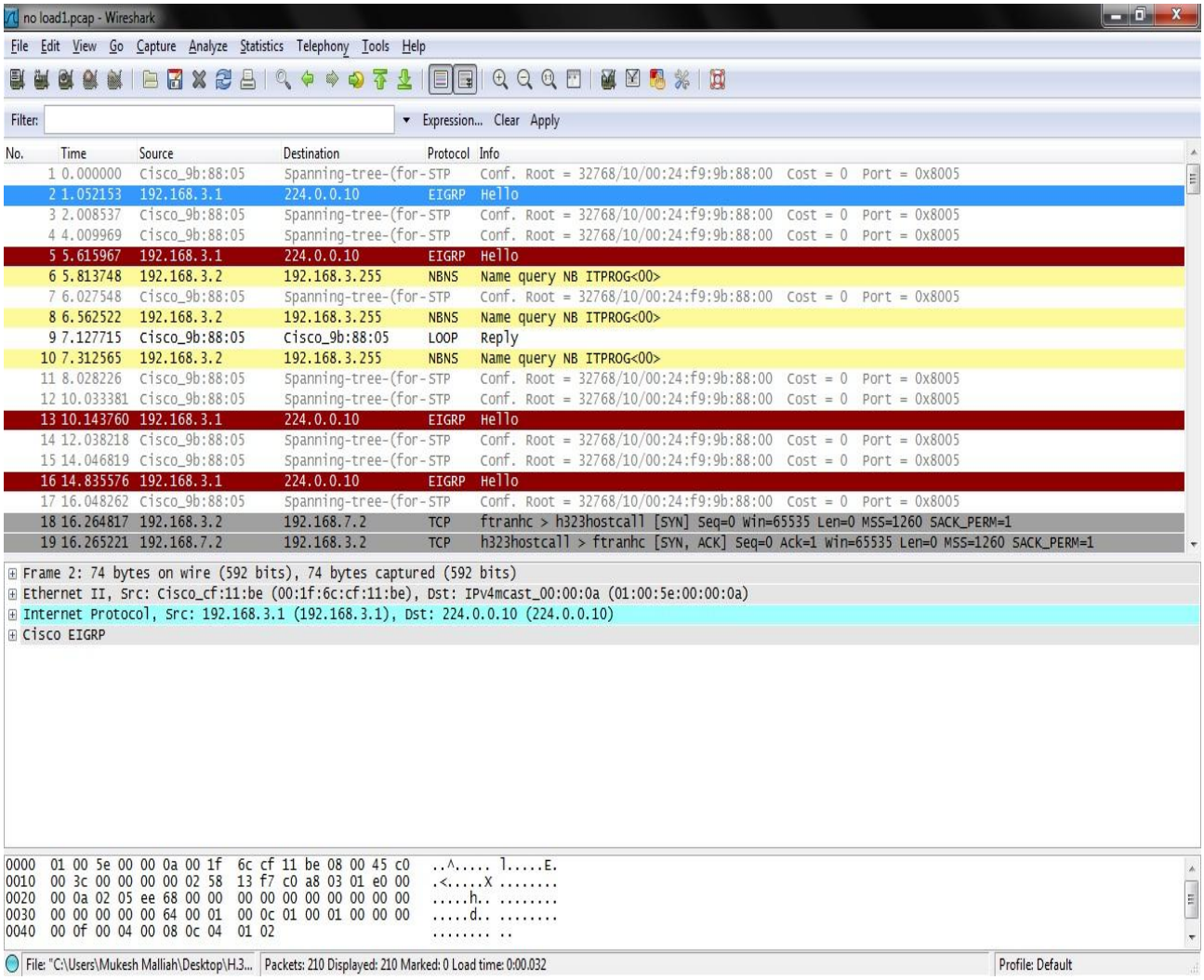
no load1.pcap - Wireshark
File Edit View Go Capture Analyze Statistics Telephony Tools Help
Filter: Expression... Clear Apply
No. Time Source Destination Protocol Info
1 0.000000 Cisco_9b:88:05 Spanning-tree-(for-STP Conf. Root = 32768/10/00:24:f9:9b:88:00 Cost = 0 Port = 0x8005
2 1.052153 192.168.3.1 224.0.0.10 EIGRP Hello
3 2.008537 Cisco_9b:88:05 Spanning-tree-(for-STP Conf. Root = 32768/10/00:24:f9:9b:88:00 Cost = 0 Port = 0x8005
4 4.009969 Cisco_9b:88:05 Spanning-tree-(for-STP Conf. Root = 32768/10/00:24:f9:9b:88:00 Cost = 0 Port = 0x8005
5 5.615967 192.168.3.1 224.0.0.10 EIGRP Hello
6 5.813748 192.168.3.2 192.168.3.255 NBNS Name query NB ITPROG<00>
7 6.027548 Cisco_9b:88:05 Spanning-tree-(for-STP Conf. Root = 32768/10/00:24:f9:9b:88:00 Cost = 0 Port = 0x8005
8 6.562522 192.168.3.2 192.168.3.255 NBNS Name query NB ITPROG<00>
9 7.127715 Cisco_9b:88:05 Cisco_9b:88:05 LOOP Reply
10 7.312565 192.168.3.2 192.168.3.255 NBNS Name query NB ITPROG<00>
11 8.028226 Cisco_9b:88:05 Spanning-tree-(for-STP Conf. Root = 32768/10/00:24:f9:9b:88:00 Cost = 0 Port = 0x8005
12 10.033381 Cisco_9b:88:05 Spanning-tree-(for-STP Conf. Root = 32768/10/00:24:f9:9b:88:00 Cost = 0 Port = 0x8005
13 10.143760 192.168.3.1 224.0.0.10 EIGRP Hello
14 12.038218 Cisco_9b:88:05 Spanning-tree-(for-STP Conf. Root = 32768/10/00:24:f9:9b:88:00 Cost = 0 Port = 0x8005
15 14.046819 Cisco_9b:88:05 Spanning-tree-(for-STP Conf. Root = 32768/10/00:24:f9:9b:88:00 Cost = 0 Port = 0x8005
16 14.835576 192.168.3.1 224.0.0.10 EIGRP Hello
17 16.048262 Cisco_9b:88:05 Spanning-tree-(for-STP Conf. Root = 32768/10/00:24:f9:9b:88:00 Cost = 0 Port = 0x8005
18 16.264817 192.168.3.2 192.168.7.2 TCP ftranhc > h323hostcall [SYN] Seq=0 Win=65535 Len=0 MSS=1260 SACK_PERM=1
19 16.265221 192.168.7.2 192.168.3.2 TCP h323hostcall > ftranhc [SYN, ACK] Seq=0 Ack=1 Win=65535 Len=0 MSS=1260 SACK_PERM=1
Frame 2: 74 bytes on wire (592 bits), 74 bytes captured (592 bits)
Ethernet II, Src: Cisco_cf:11:be (00:1f:6c:cf:11:be), Dst: IPv4mcast_00:00:0a (01:00:5e:00:00:0a)
Internet Protocol, Src: 192.168.3.1 (192.168.3.1), Dst: 224.0.0.10 (224.0.0.10)
Cisco EIGRP
0000 01 00 5e 00 00 0a 00 1f 6c cf 11 be 08 00 45 c0 ..^..... l.....E.
0010 00 3c 00 00 00 00 02 58 13 f7 c0 a8 03 01 e0 00 .<.....X ....
0020 00 0a 02 05 ee 68 00 00 00 00 00 00 00 00 00 00 .....h.. ....
0030 00 00 00 00 00 64 00 01 00 0c 01 00 01 00 00 00 .....d.. ....
0040 00 0f 00 04 00 08 0c 04 01 02 ....
File: "C:\Users\Mukesh Malliah\Desktop\H.3... Packets: 210 Displayed: 210 Marked: 0 Load time: 0:00.032
Profile: Default

Fig (5.1) Active functions of H.323 network topology

5.2 Traffic Generated

In order to analyse the network topology on a real time status, a traffic generator is used to generate the traffic of various types over the network. The traffic generator is linked with the data computer through a switch over Vlan 10. Hence it can generate traffic from the computer connected to Vlan 10. The figure 5.2 shown below clearly mentions all kinds of traffics like

FTP, Telnet, NTP, SMTP and POP messages are generated in the network along with our voice packets which transferred through RTP. The packet highlighted in blue colour represents the voice packet and the remaining are the various types of data packets produced by the traffic generator.

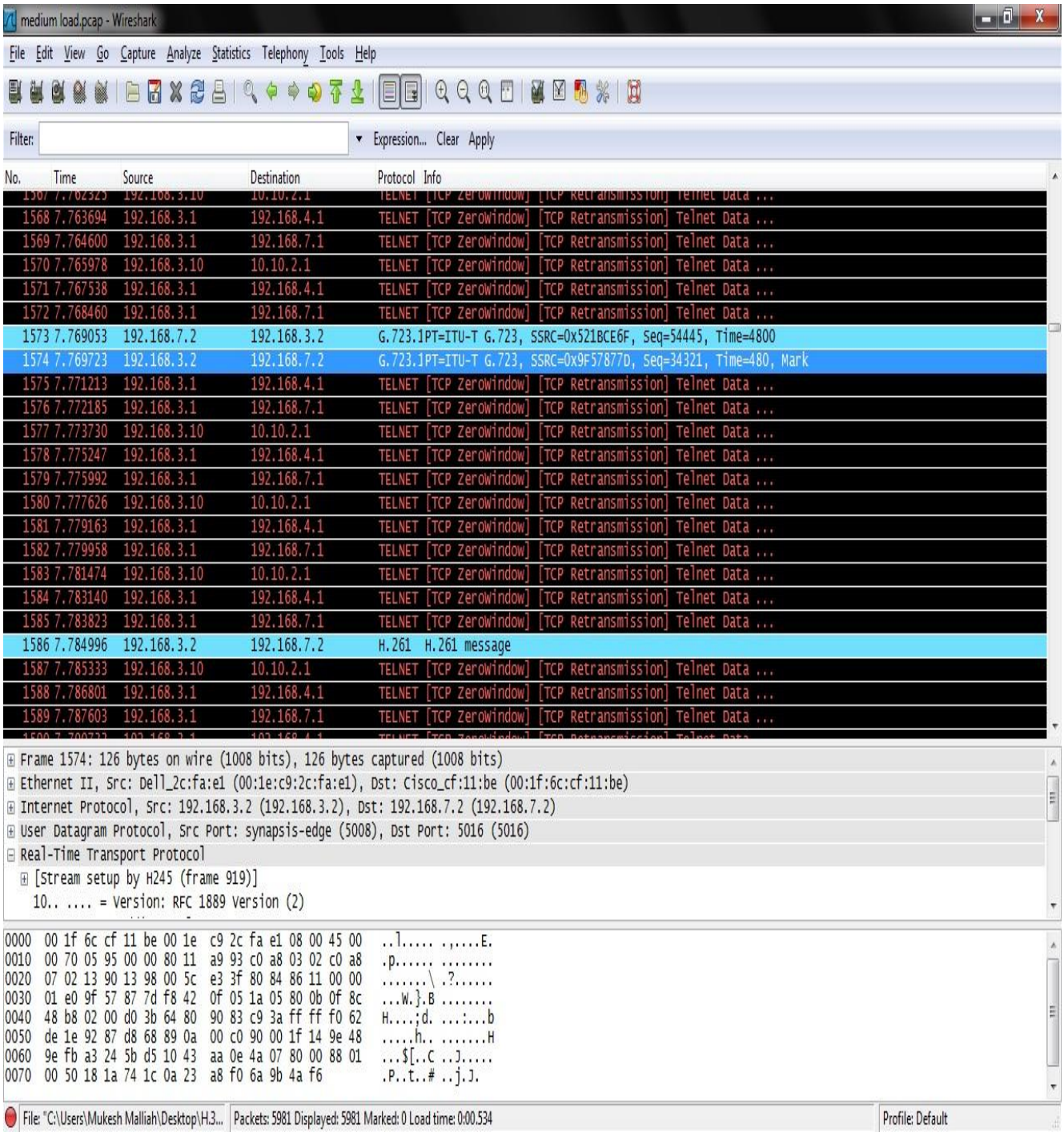
medium load.pcap - Wireshark
File Edit View Go Capture Analyze Statistics Telephony Tools Help
Filter: Expression... Clear Apply
No. Time Source Destination Protocol Info
1568 7.763694 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1569 7.764600 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1570 7.765978 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1571 7.767538 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1572 7.768460 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1573 7.769053 192.168.7.2 192.168.3.2 G.723.1PT=ITU-T G.723, SSRC=0x521BCE6F, Seq=54445, Time=4800
1574 7.769723 192.168.3.2 192.168.7.2 G.723.1PT=ITU-T G.723, SSRC=0x9F57877D, Seq=34321, Time=480, Mark
1575 7.771213 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1576 7.772185 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1577 7.773730 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1578 7.775247 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1579 7.775992 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1580 7.777626 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1581 7.779163 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1582 7.779958 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1583 7.781474 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1584 7.783140 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1585 7.783823 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1586 7.784996 192.168.3.2 192.168.7.2 H.261 H.261 message
1587 7.785333 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1588 7.786801 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1589 7.787603 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
Frame 1574: 126 bytes on wire (1008 bits), 126 bytes captured (1008 bits)
Ethernet II, Src: Dell_2c:fa:e1 (00:1e:c9:2c:fa:e1), Dst: Cisco_cf:11:be (00:1f:6c:cf:11:be)
Internet Protocol, Src: 192.168.3.2 (192.168.3.2), Dst: 192.168.7.2 (192.168.7.2)
User Datagram Protocol, Src Port: synapsis-edge (5008), Dst Port: 5016 (5016)
Real-Time Transport Protocol
[Stream setup by H245 (frame 919)]
10.. .... = Version: RFC 1889 Version (2)
0000 00 1f 6c cf 11 be 00 1e c9 2c fa e1 08 00 45 00
0010 00 70 05 95 00 00 80 11 a9 93 c0 a8 03 02 c0 a8
0020 07 02 13 90 13 98 00 5c e3 3f 80 84 86 11 00 00
0030 01 e0 9f 57 87 7d f8 42 0f 05 1a 05 80 0b 0f 8c
0040 48 b8 02 00 d0 3b 64 80 90 83 c9 3a ff ff f0 62
0050 de 1e 92 87 d8 68 89 0a 00 c0 90 00 1f 14 9e 48
0060 9e fb a3 24 5b d5 10 43 aa 0e 4a 07 80 00 88 01
0070 00 50 18 1a 74 1c 0a 23 a8 f0 6a 9b 4a f6
File: "C:\Users\Mukesh Malliah\Desktop\H3... Packets: 5981 Displayed: 5981 Marked: 0 Load time: 0:00.534 Profile: Default

Fig (5.2) Traffic generated in H.323 network topology

5.3 Call setup at various loads

5.3.1 No Load

The first experiment in this call setup delay analyses is performed under ZERO load. In this process only the voice packets were transmitted through the network and the traffic generator was kept OFF. The voice packets were transmitted between the Cisco IP soft phone and the traditional phone connected to the PBX. To setup a call between these two terminals it took almost 177 milliseconds. We declare this time from the average obtained by establishing a number of calls under the same condition. The fig 5.3.1 is an example to show the experiment and highlighted circle defines the call setup time. In that mean time, the H.323 phone and traditional phones are in the process of establishing the call by sending TCP and H.323 messages.

Here the messages transferred to connect the call are, Call State Message, Call Info Message, Gatekeeper registration, Gatekeeper admission, Gatekeeper location, Open Receive Channel, Open Receive channel Acknowledgement and Start Media Transmission are the messages transferred by H.323 and REQ and ACK are done by TCP to setup the calls.

Call No	**Calls Made Between**	**Process Time in Chart**	**Call setup delay (ms)**
Call 1	5551-5559	16.264-16.437	173
Call 2	5551-5556001	17.748-17.924	176
Call 3	5559-5551	15.257-15.531	194
Call 4	5556001-5551	16.203-16.374	171
Call 5	5552-5556001	15.881-16.067	186
Call 6	5552-5559	16.786-16.953	167
Call 7	5559-5552	15.632-15.801	169
Call 8	5556001-5552	16.849-17.034	185
Call 9	5551-5556001	15.187-15.358	171
Call 10	5552-5559	16.503-16.678	175
			Arithmetic Mean = 177ms **Standard Deviation = 8ms**

Tab (5.3.1) Call Setup delay of H.323 at no load

Arithmetic mean A = $\sum$ (call setup delay)/N

= (173+176+194+171+186+167+169+185+171+175)/10

A =177 ms

Standard Deviation S = $\sqrt{}$ ((call setup delay- Arithmetic Mean) ^2)/N

S = $\sqrt{}$ (16+1+289+36+81+100+64+64+36+4)/10

= $\sqrt{}$691/10

S = 8 ms

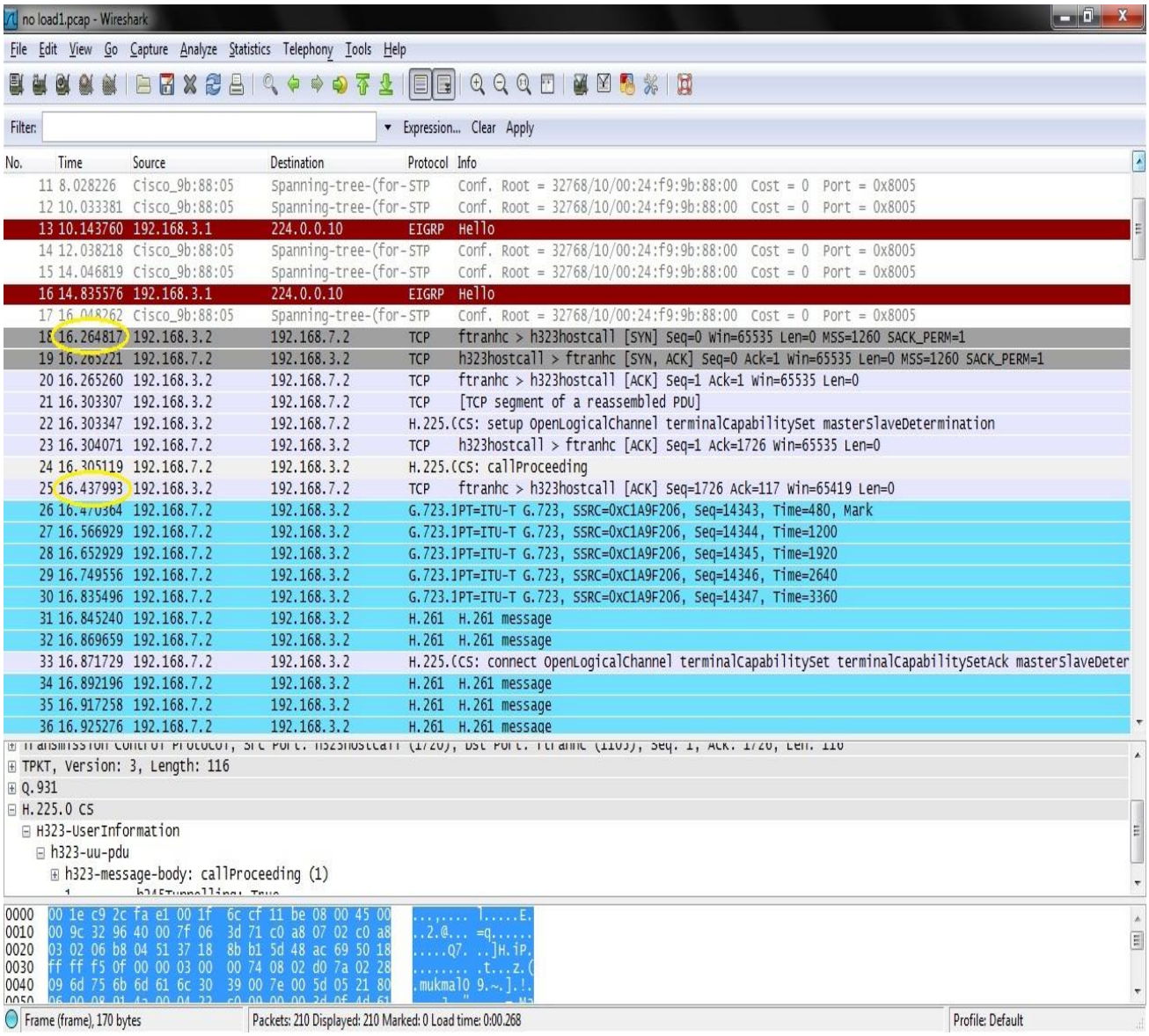

Fig (5.3.1) Call Setup delay of H.323 at no load

5.3.2 Medium Load

The fig 5.3.2a and 5.3.2b shown below represents the second experiment and which was undertaken under a medium load. In order to do the traffic generator was kept ON and it was assigned to a load of 200. Before setting up the call between the terminals the traffic generator was started first and then the call has been made to analyze the call setup time under this medium load. Call setup time is estimated by connecting a number of calls under the same condition and the mean time in setting the call is estimated around 253 milliseconds. The fig 5.3.2a shows the initiation of the call setup time and it is highlighted at 7.011th second. And the fig 5.3.2c shows the end of the call setup process and is noted at 7.221th second. The time interval of this process takes almost 210 milliseconds. The call mentioned in the figure is an example to explain the process among the number of calls made to estimate call setup time.

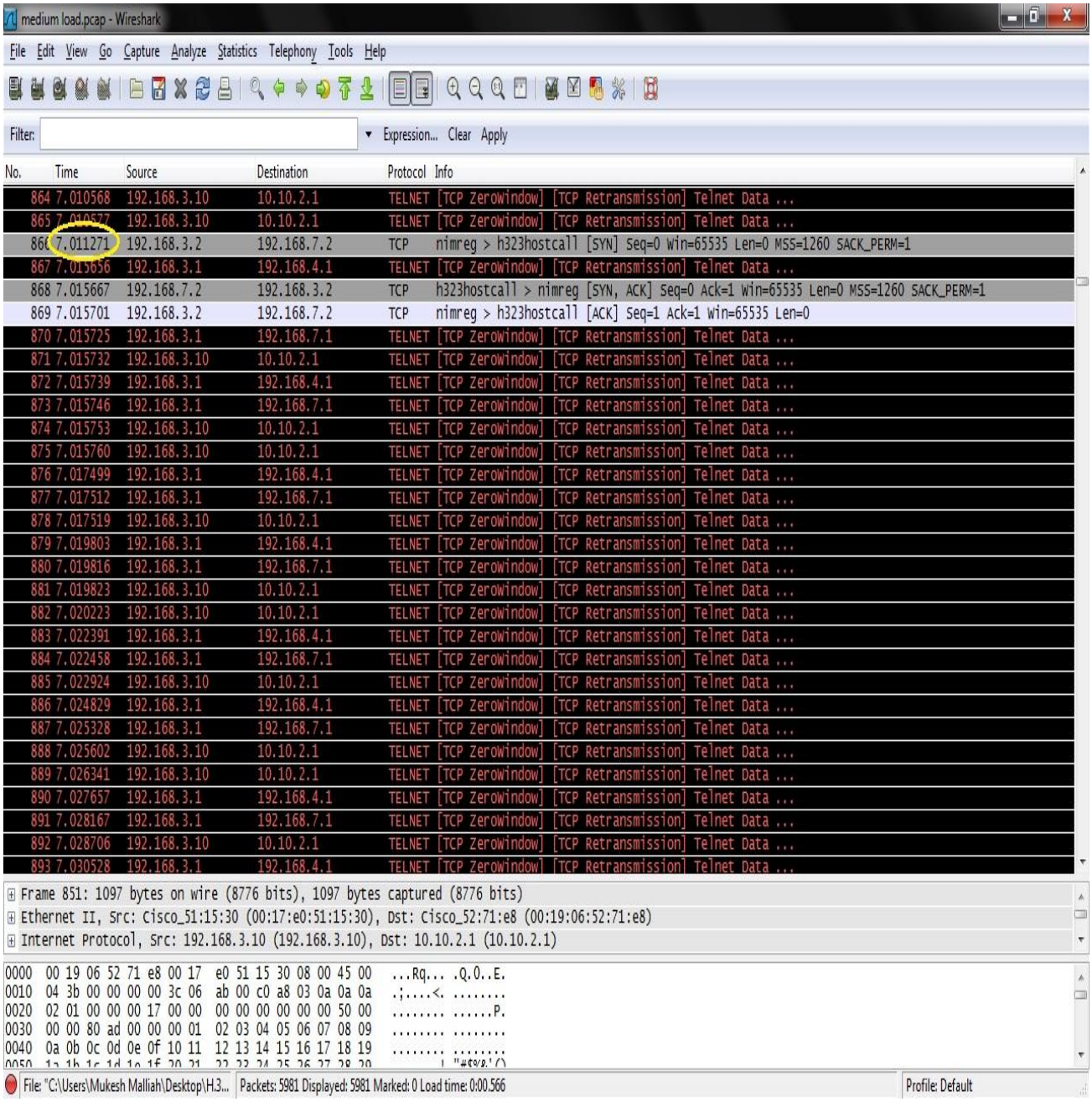
medium load.pcap - Wireshark
File Edit View Go Capture Analyze Statistics Telephony Tools Help
Filter: Expression... Clear Apply
No. Time Source Destination Protocol Info
864 7.010568 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
865 7.010577 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
866 7.011271 192.168.3.2 192.168.7.2 TCP nimreg > h323hostcall [SYN] Seq=0 Win=65535 Len=0 MSS=1260 SACK_PERM=1
867 7.015656 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
868 7.015667 192.168.7.2 192.168.3.2 TCP h323hostcall > nimreg [SYN, ACK] Seq=0 Ack=1 Win=65535 Len=0 MSS=1260 SACK_PERM=1
869 7.015701 192.168.3.2 192.168.7.2 TCP nimreg > h323hostcall [ACK] Seq=1 Ack=1 Win=65535 Len=0
870 7.015725 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
871 7.015732 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
872 7.015739 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
873 7.015746 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
874 7.015753 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
875 7.015760 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
876 7.017499 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
877 7.017512 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
878 7.017519 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
879 7.019803 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
880 7.019816 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
881 7.019823 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
882 7.020223 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
883 7.022391 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
884 7.022458 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
885 7.022924 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
886 7.024829 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
887 7.025328 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
888 7.025602 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
889 7.026341 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
890 7.027657 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
891 7.028167 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
892 7.028706 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
893 7.030528 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
Frame 851: 1097 bytes on wire (8776 bits), 1097 bytes captured (8776 bits)
Ethernet II, Src: Cisco_51:15:30 (00:17:e0:51:15:30), Dst: Cisco_52:71:e8 (00:19:06:52:71:e8)
Internet Protocol, Src: 192.168.3.10 (192.168.3.10), Dst: 10.10.2.1 (10.10.2.1)
0000 00 19 06 52 71 e8 00 17 e0 51 15 30 08 00 45 00 ...Rq... .Q.0..E.
0010 04 3b 00 00 00 00 3c 06 ab 00 c0 a8 03 0a 0a 0a .;....<. ....
0020 02 01 00 00 00 17 00 00 00 00 00 00 00 00 50 00 ....P.
0030 00 00 80 ad 00 00 00 01 02 03 04 05 06 07 08 09 ....
0040 0a 0b 0c 0d 0e 0f 10 11 12 13 14 15 16 17 18 19 ....
File: "C:\Users\Mukesh Malliah\Desktop\H.3... Packets: 5981 Displayed: 5981 Marked: 0 Load time: 0:00.566
Profile: Default

Fig (5.3.2a) Start time to set a call in H.323 at normal load

Call No	Calls Made Between	Process Time in Chart	Call setup delay(ms)
Call 1	5551-5559	7.011-7.221	210
Call 2	5551-5556001	12.416-12.681	265
Call 3	5559-5551	21.732-21.976	244
Call 4	5556001-5551	32.739-33.034	295
Call 5	5552-5556001	16.394-16.603	209

Call 6	5552-5559	23.487-23.751	264
Call 7	5559-5552	20.043-20.311	268
Call 8	5556001-5552	21.300-21.561	261
Call 9	5551-5556001	16.049-16.294	245
Call 10	5552-5559	22.602-22.873	271
			Arithmetic Mean = 253 ms **Standard Deviation = 26 ms**

Tab (5.3.2) Call Setup delays of H.323 at medium loads

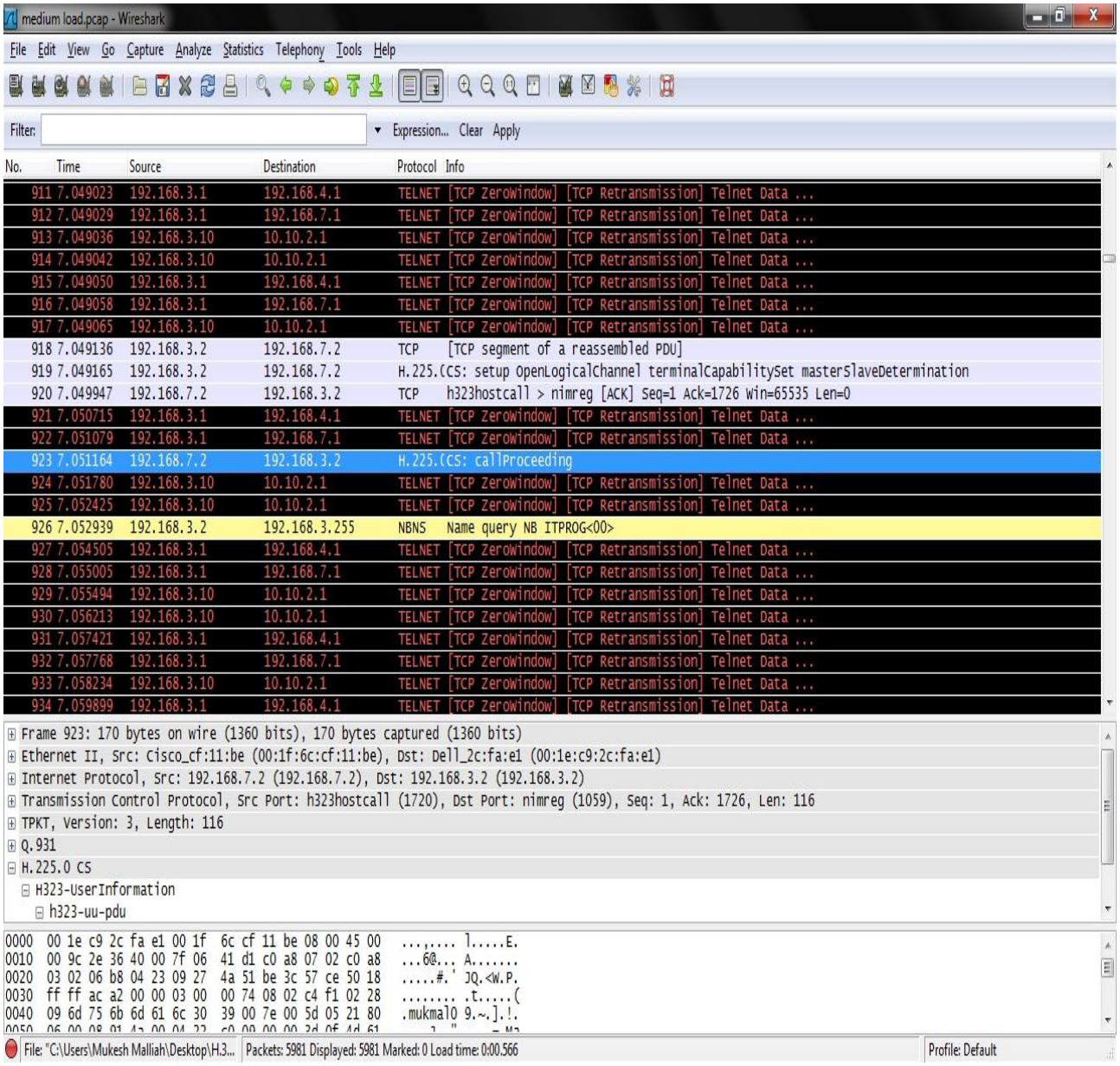
medium load.pcap - Wireshark
File Edit View Go Capture Analyze Statistics Telephony Tools Help
Filter: Expression... Clear Apply
No. Time Source Destination Protocol Info
911 7.049023 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
912 7.049029 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
913 7.049036 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
914 7.049042 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
915 7.049050 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
916 7.049058 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
917 7.049065 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
918 7.049136 192.168.3.2 192.168.7.2 TCP [TCP segment of a reassembled PDU]
919 7.049165 192.168.3.2 192.168.7.2 H.225.(CS: setup OpenLogicalChannel terminalCapabilitySet masterSlaveDetermination
920 7.049947 192.168.7.2 192.168.3.2 TCP h323hostcall > nimreg [ACK] Seq=1 Ack=1726 Win=65535 Len=0
921 7.050715 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
922 7.051079 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
923 7.051164 192.168.7.2 192.168.3.2 H.225.(CS: callProceeding
924 7.051780 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
925 7.052425 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
926 7.052939 192.168.3.2 192.168.3.255 NBNS Name query NB ITPROG<00>
927 7.054505 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
928 7.055005 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
929 7.055494 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
930 7.056213 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
931 7.057421 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
932 7.057768 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
933 7.058234 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
934 7.059899 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
Frame 923: 170 bytes on wire (1360 bits), 170 bytes captured (1360 bits)
Ethernet II, Src: Cisco_cf:11:be (00:1f:6c:cf:11:be), Dst: Dell_2c:fa:e1 (00:1e:c9:2c:fa:e1)
Internet Protocol, Src: 192.168.7.2 (192.168.7.2), Dst: 192.168.3.2 (192.168.3.2)
Transmission Control Protocol, Src Port: h323hostcall (1720), Dst Port: nimreg (1059), Seq: 1, Ack: 1726, Len: 116
TPKT, Version: 3, Length: 116
Q.931
H.225.0 CS
H323-UserInformation
h323-uu-pdu
0000 00 1e c9 2c fa e1 00 1f 6c cf 11 be 08 00 45 00 ...,.... l.....E.
0010 00 9c 2e 36 40 00 7f 06 41 d1 c0 a8 07 02 c0 a8 ...6@... A.....
0020 03 02 06 b8 04 23 09 27 4a 51 be 3c 57 ce 50 18 .....#.' JQ.<W.P.
0030 ff ff ac a2 00 00 03 00 00 74 08 02 c4 f1 02 28 ..... .t.....(
0040 09 6d 75 6b 6d 61 6c 30 39 00 7e 00 5d 05 21 80 .mukmal0 9.~.].!.
File: "C:\Users\Mukesh Malliah\Desktop\H.3... Packets: 5981 Displayed: 5981 Marked: 0 Load time: 0:00.566
Profile: Default

Fig (5.3.2b) Call connecting process of H.323 at normal load

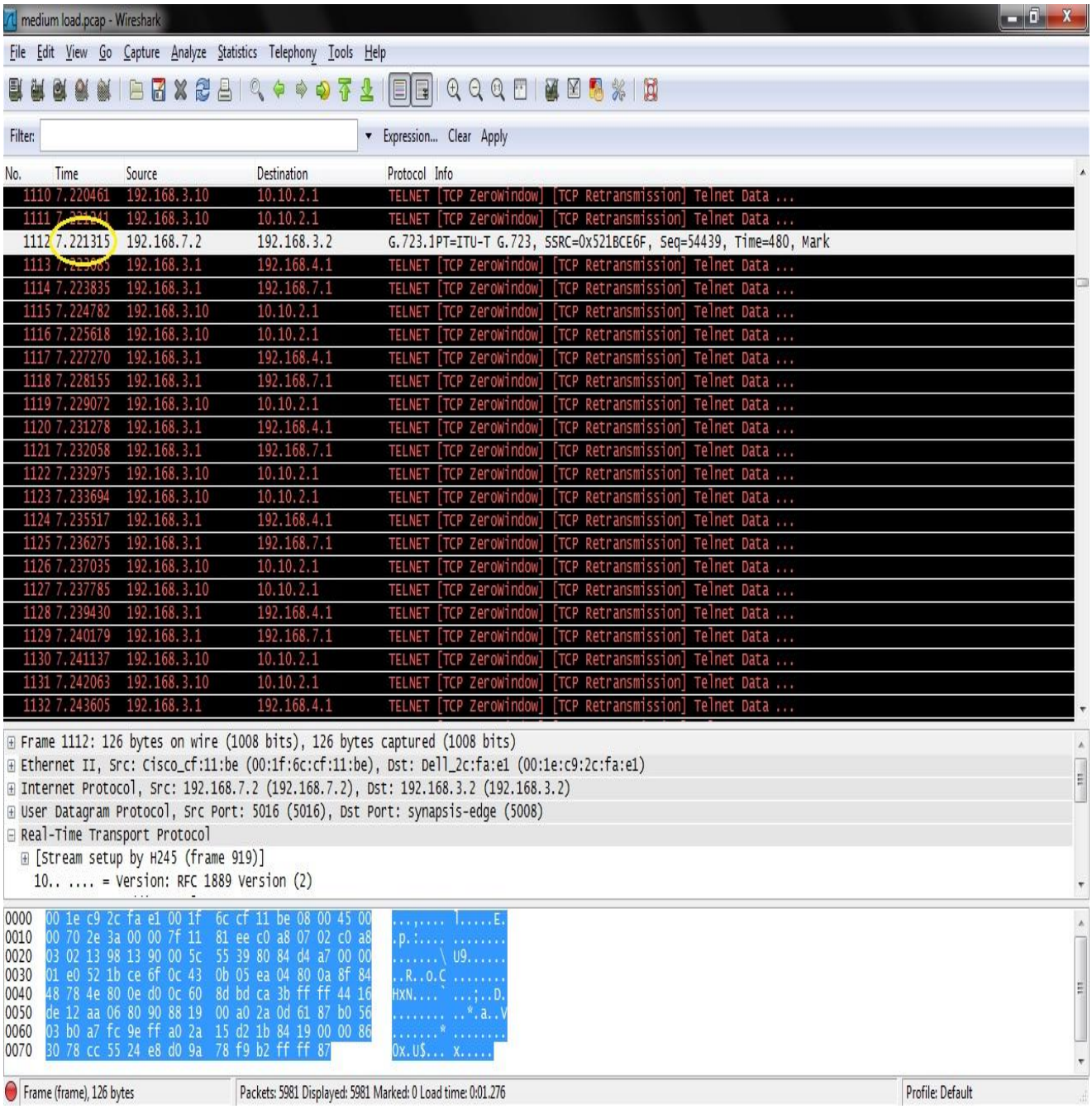
medium load.pcap - Wireshark
File Edit View Go Capture Analyze Statistics Telephony Tools Help
Filter: Expression... Clear Apply
No. Time Source Destination Protocol Info
1110 7.220461 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1111 7.221241 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1112 7.221315 192.168.7.2 192.168.3.2 G.723.1PT=ITU-T G.723, SSRC=0x521BCE6F, Seq=54439, Time=480, Mark
1113 7.223083 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1114 7.223835 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1115 7.224782 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1116 7.225618 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1117 7.227270 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1118 7.228155 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1119 7.229072 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1120 7.231278 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1121 7.232058 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1122 7.232975 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1123 7.233694 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1124 7.235517 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1125 7.236275 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1126 7.237035 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1127 7.237785 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1128 7.239430 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1129 7.240179 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1130 7.241137 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1131 7.242063 192.168.3.10 10.10.2.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
1132 7.243605 192.168.3.1 192.168.4.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
Frame 1112: 126 bytes on wire (1008 bits), 126 bytes captured (1008 bits)
Ethernet II, Src: Cisco_cf:11:be (00:1f:6c:cf:11:be), Dst: Dell_2c:fa:e1 (00:1e:c9:2c:fa:e1)
Internet Protocol, Src: 192.168.7.2 (192.168.7.2), Dst: 192.168.3.2 (192.168.3.2)
User Datagram Protocol, Src Port: 5016 (5016), Dst Port: synapsis-edge (5008)
Real-Time Transport Protocol
[Stream setup by H245 (frame 919)]
10.. .... = Version: RFC 1889 Version (2)
0000 00 1e c9 2c fa e1 00 1f 6c cf 11 be 08 00 45 00
0010 00 70 2e 3a 00 00 7f 11 81 ee c0 a8 07 02 c0 a8
0020 03 02 13 98 13 90 00 5c 55 39 80 84 d4 a7 00 00
0030 01 e0 52 1b ce 6f 0c 43 0b 05 ea 04 80 0a 8f 84
0040 48 78 4e 80 0e d0 0c 60 8d bd ca 3b ff ff 44 16
0050 de 12 aa 06 80 90 88 19 00 a0 2a 0d 61 87 b0 56
0060 03 b0 a7 fc 9e ff a0 2a 15 d2 1b 84 19 00 00 86
0070 30 78 cc 55 24 e8 d0 9a 78 f9 b2 ff ff 87
Frame (frame), 126 bytes
Packets: 5981 Displayed: 5981 Marked: 0 Load time: 0:01.276
Profile: Default

Fig (5.3.2c) Call connected time of H.323 at normal load

Arithmetic mean A = ∑ (call setup delay)/N

A = 253ms

Standard Deviation S = √ ((call setup delay- Arithmetic Mean) ^2)/N

S = 26 ms

5.3.3 Heavy Load

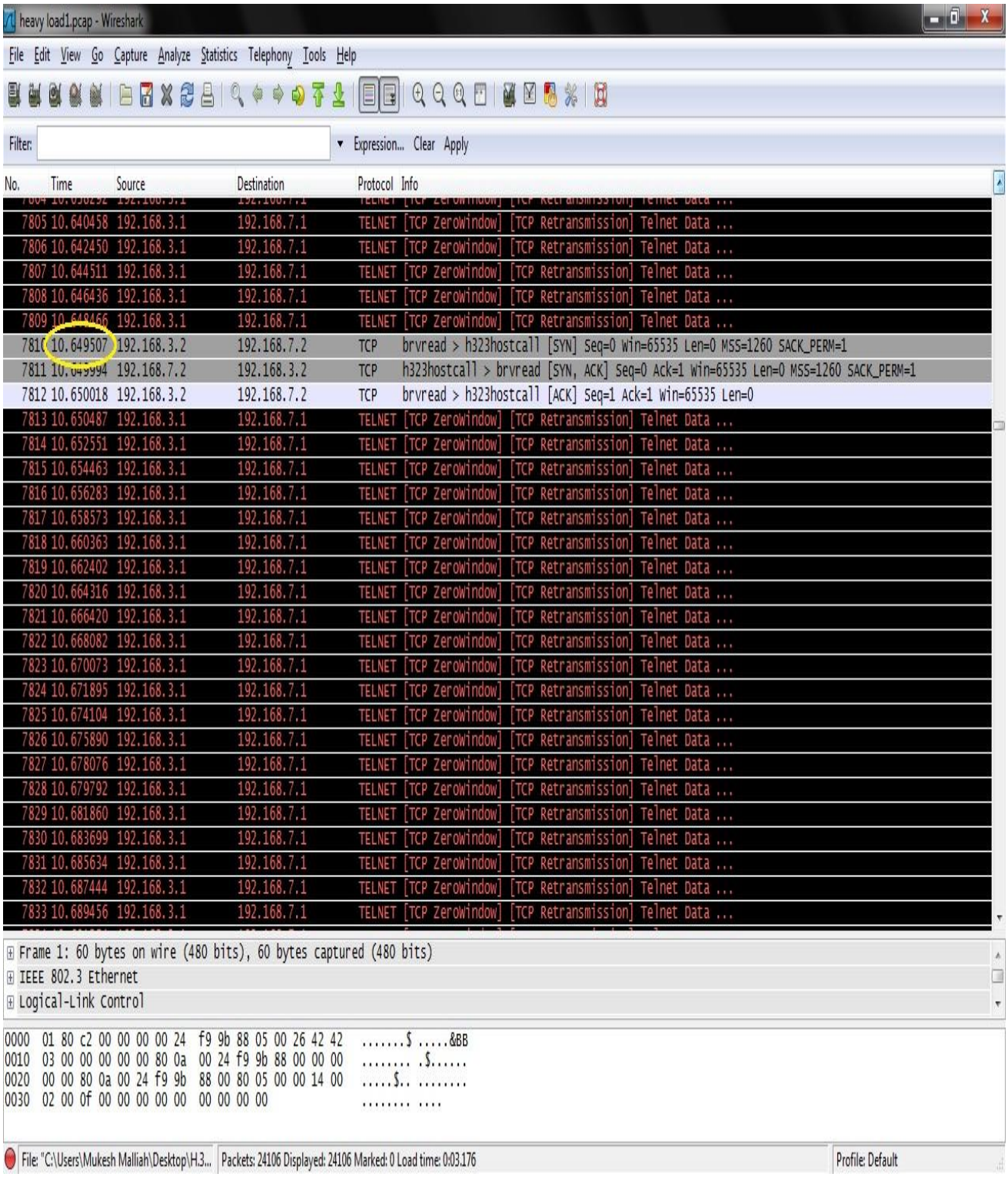
heavy load1.pcap - Wireshark
File Edit View Go Capture Analyze Statistics Telephony Tools Help
Filter: Expression... Clear Apply
No. Time Source Destination Protocol Info
7805 10.640458 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7806 10.642450 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7807 10.644511 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7808 10.646436 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7809 10.648466 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7810 10.649507 192.168.3.2 192.168.7.2 TCP brvread > h323hostcall [SYN] Seq=0 Win=65535 Len=0 MSS=1260 SACK_PERM=1
7811 10.649994 192.168.7.2 192.168.3.2 TCP h323hostcall > brvread [SYN, ACK] Seq=0 Ack=1 Win=65535 Len=0 MSS=1260 SACK_PERM=1
7812 10.650018 192.168.3.2 192.168.7.2 TCP brvread > h323hostcall [ACK] Seq=1 Ack=1 Win=65535 Len=0
7813 10.650487 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7814 10.652551 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7815 10.654463 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7816 10.656283 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7817 10.658573 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7818 10.660363 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7819 10.662402 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7820 10.664316 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7821 10.666420 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7822 10.668082 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7823 10.670073 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7824 10.671895 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7825 10.674104 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7826 10.675890 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7827 10.678076 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7828 10.679792 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7829 10.681860 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7830 10.683699 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7831 10.685634 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7832 10.687444 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7833 10.689456 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
Frame 1: 60 bytes on wire (480 bits), 60 bytes captured (480 bits)
IEEE 802.3 Ethernet
Logical-Link Control
0000 01 80 c2 00 00 00 00 24 f9 9b 88 05 00 26 42 42 ....$ .....&BB
0010 03 00 00 00 00 00 80 0a 00 24 f9 9b 88 00 00 00 ....$.....
0020 00 00 80 0a 00 24 f9 9b 88 00 80 05 00 00 14 00 .....$.. ....
0030 02 00 0f 00 00 00 00 00 00 00 00 00 ....
File: "C:\Users\Mukesh Malliah\Desktop\H.3... Packets: 24106 Displayed: 24106 Marked: 0 Load time: 0:03.176 Profile: Default

Fig (5.3.3a) Start time to set a call in H.323 at heavy load

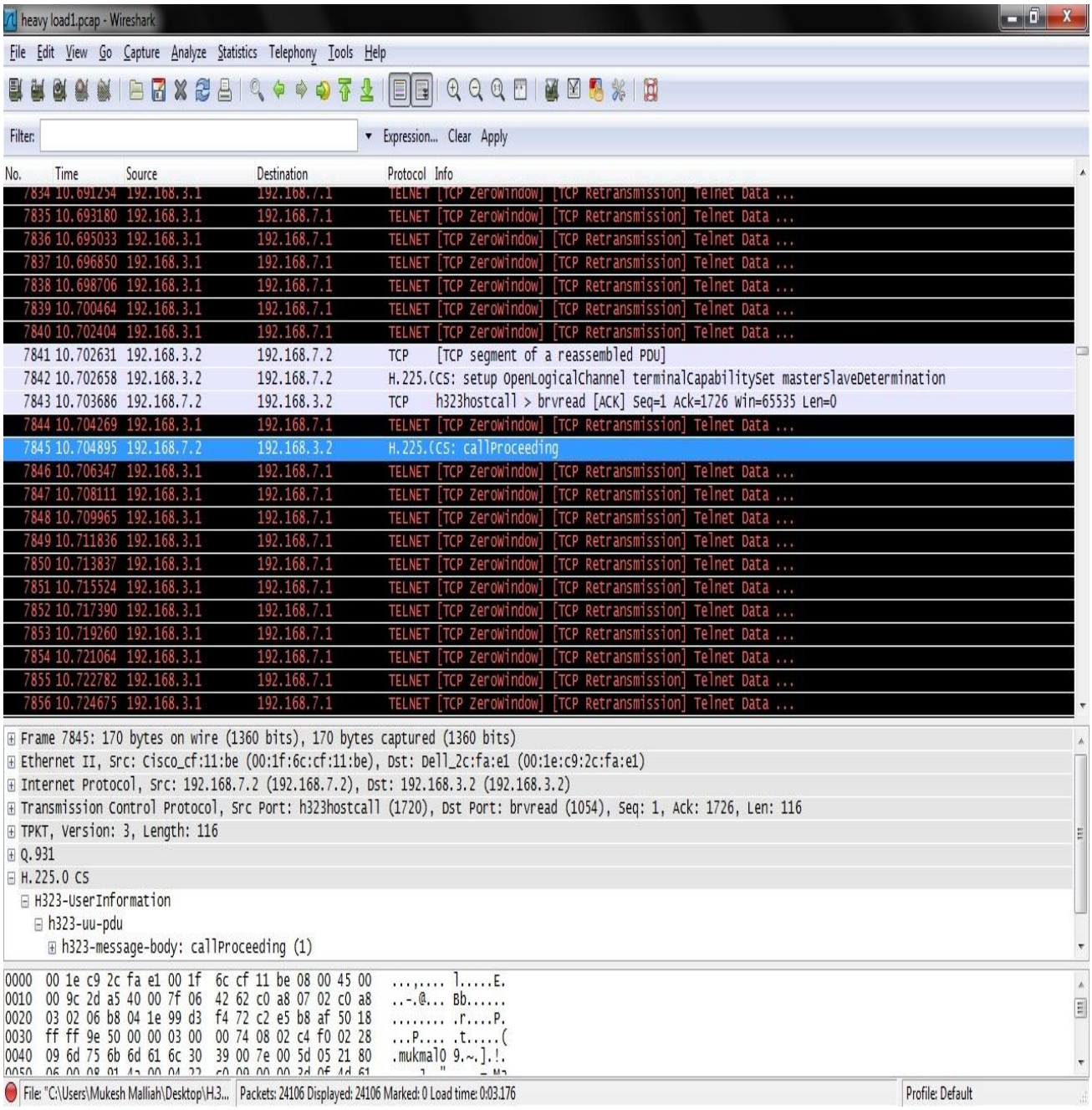
heavy load1.pcap - Wireshark
File Edit View Go Capture Analyze Statistics Telephony Tools Help
Filter: Expression... Clear Apply
No. Time Source Destination Protocol Info
7834 10.691254 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7835 10.693180 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7836 10.695033 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7837 10.696850 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7838 10.698706 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7839 10.700464 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7840 10.702404 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7841 10.702631 192.168.3.2 192.168.7.2 TCP [TCP segment of a reassembled PDU]
7842 10.702658 192.168.3.2 192.168.7.2 H.225.(CS: setup OpenLogicalChannel terminalCapabilitySet masterSlaveDetermination
7843 10.703686 192.168.7.2 192.168.3.2 TCP h323hostcall > brvread [ACK] Seq=1 Ack=1726 Win=65535 Len=0
7844 10.704269 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7845 10.704895 192.168.7.2 192.168.3.2 H.225.(CS: callProceeding
7846 10.706347 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7847 10.708111 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7848 10.709965 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7849 10.711836 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7850 10.713837 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7851 10.715524 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7852 10.717390 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7853 10.719260 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7854 10.721064 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7855 10.722782 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7856 10.724675 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
Frame 7845: 170 bytes on wire (1360 bits), 170 bytes captured (1360 bits)
Ethernet II, Src: Cisco_cf:11:be (00:1f:6c:cf:11:be), Dst: Dell_2c:fa:e1 (00:1e:c9:2c:fa:e1)
Internet Protocol, Src: 192.168.7.2 (192.168.7.2), Dst: 192.168.3.2 (192.168.3.2)
Transmission Control Protocol, Src Port: h323hostcall (1720), Dst Port: brvread (1054), Seq: 1, Ack: 1726, Len: 116
TPKT, Version: 3, Length: 116
Q.931
H.225.0 CS
H323-UserInformation
h323-uu-pdu
h323-message-body: callProceeding (1)
0000 00 1e c9 2c fa e1 00 1f 6c cf 11 be 08 00 45 00 ...,.... l.....E.
0010 00 9c 2d a5 40 00 7f 06 42 62 c0 a8 07 02 c0 a8 ..-.@... Bb......
0020 03 02 06 b8 04 1e 99 d3 f4 72 c2 e5 b8 af 50 18 ..... .r....P.
0030 ff ff 9e 50 00 00 03 00 00 74 08 02 c4 f0 02 28 ...P.... .t.....(
0040 09 6d 75 6b 6d 61 6c 30 39 00 7e 00 5d 05 21 80 .mukmal0 9.~.].!.
File: "C:\Users\Mukesh Malliah\Desktop\H.3... Packets: 24106 Displayed: 24106 Marked: 0 Load time: 0:03.176
Profile: Default

Fig (5.3.3b) Call connecting process of H.323 at heavy load

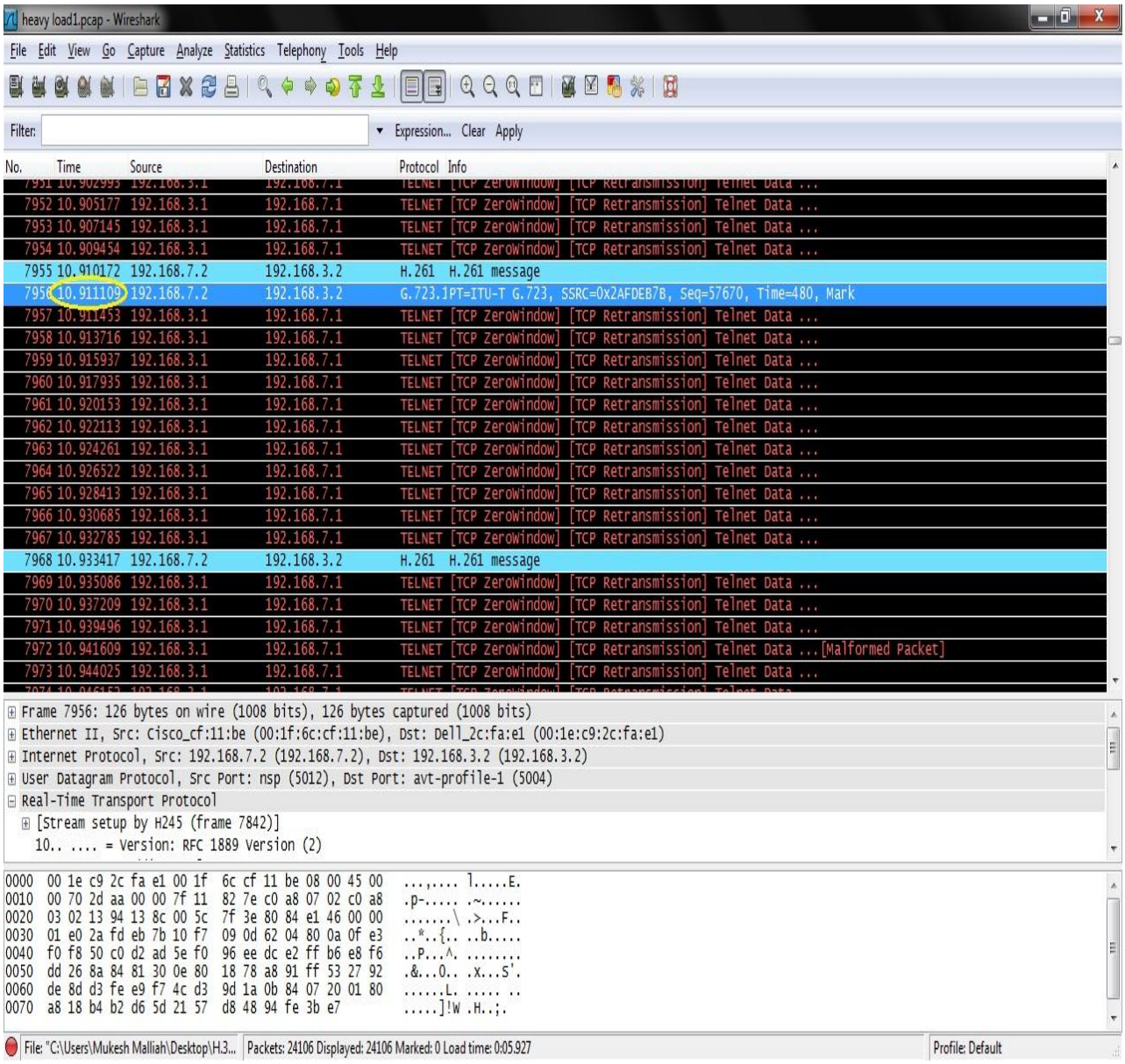
heavy load1.pcap - Wireshark
File Edit View Go Capture Analyze Statistics Telephony Tools Help
Filter: Expression... Clear Apply
No. Time Source Destination Protocol Info
7952 10.905177 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7953 10.907145 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7954 10.909454 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7955 10.910172 192.168.7.2 192.168.3.2 H.261 H.261 message
7956 10.911109 192.168.7.2 192.168.3.2 G.723.1PT=ITU-T G.723, SSRC=0x2AFDEB7B, Seq=57670, Time=480, Mark
7957 10.911453 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7958 10.913716 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7959 10.915937 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7960 10.917935 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7961 10.920153 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7962 10.922113 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7963 10.924261 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7964 10.926522 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7965 10.928413 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7966 10.930685 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7967 10.932785 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7968 10.933417 192.168.7.2 192.168.3.2 H.261 H.261 message
7969 10.935086 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7970 10.937209 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7971 10.939496 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
7972 10.941609 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...[Malformed Packet]
7973 10.944025 192.168.3.1 192.168.7.1 TELNET [TCP ZeroWindow] [TCP Retransmission] Telnet Data ...
Frame 7956: 126 bytes on wire (1008 bits), 126 bytes captured (1008 bits)
Ethernet II, Src: Cisco_cf:11:be (00:1f:6c:cf:11:be), Dst: Dell_2c:fa:e1 (00:1e:c9:2c:fa:e1)
Internet Protocol, Src: 192.168.7.2 (192.168.7.2), Dst: 192.168.3.2 (192.168.3.2)
User Datagram Protocol, Src Port: nsp (5012), Dst Port: avt-profile-1 (5004)
Real-Time Transport Protocol
[Stream setup by H245 (frame 7842)]
10.. .... = Version: RFC 1889 Version (2)
0000 00 1e c9 2c fa e1 00 1f 6c cf 11 be 08 00 45 00 ...,.... l.....E.
0010 00 70 2d aa 00 00 7f 11 82 7e c0 a8 07 02 c0 a8 .p-..... .~......
0020 03 02 13 94 13 8c 00 5c 7f 3e 80 84 e1 46 00 00 ....\ .>...F..
0030 01 e0 2a fd eb 7b 10 f7 09 0d 62 04 80 0a 0f e3 ..*..{.. ..b.....
0040 f0 f8 50 c0 d2 ad 5e f0 96 ee dc e2 ff b6 e8 f6 ..P...^. ....
0050 dd 26 8a 84 81 30 0e 80 18 78 a8 91 ff 53 27 92 .&...0.. .x...S'.
0060 de 8d d3 fe e9 f7 4c d3 9d 1a 0b 84 07 20 01 80 ......L. ....
0070 a8 18 b4 b2 d6 5d 21 57 d8 48 94 fe 3b e7 .....]!W .H..;.
File: "C:\Users\Mukesh Malliah\Desktop\H.3... Packets: 24106 Displayed: 24106 Marked: 0 Load time: 0:05.927 Profile: Default

Fig (5.3.3c) Call connected time of H.323 at heavy load

The final step to analyze the call setup time is also performed by making number of calls under a heavy load of traffic and from the average of call setup time of each call; the call setup delay is estimated as 360 milliseconds. Here the traffic generator was set to 1000 under the same bandwidth and then the call setup was executed. The fig 5.3.3a, 5.3.3b and 5.3.3c are the examples of the call setup process made under heavy traffic and the call initiation time here is noted at 10.649th second and from the fig 5.3.3c, the time for the final step of the call setup is noted as 10.911th second. From the results of one of the calls, the time interval between them to setup a call is almost 262 milliseconds under a heavy traffic network.

Call No	Calls Made Between	Process Time in Chart	Call setup delay(ms)
Call 1	5551-5556001	10.649-10.911	262
Call 2	5551-5559	14.346-14.687	341
Call 3	5559-5551	16.060-16.359	299
Call 4	5556001-5551	28.301-28.698	397
Call 5	5552-5556001	17.918-18.327	409
Call 6	5552-5559	23.266-23.654	388
Call 7	5559-5552	19.459-19.876	417
Call 8	5556001-5552	23.919-24.211	292
Call 9	5551-5556001	21.194-21.587	393
Call 10	5552-5559	16.994-17.398	404
			Arithmetic Mean = 360 ms **Standard Deviation = 54ms**

Tab (5.3.3) Call Setup delays of H.323 at heavy loads

Arithmetic mean A = ∑ (call setup delay)/N
A = 360 ms
Standard Deviation S = √ ((call setup delay- Arithmetic Mean) ^2)/N
S = 54 ms

5.4 Call Setup Delay results on SIP

The call setup results of H.323 are done experimentally. The SIP call setup results are derived theoretically.
The difference between a H.323 using network and the SIP using network is the Server and the transport protocol. Here in H.323, as mentioned earlier in section 4, it uses a Gatekeeper and TCP as the call setup protocol and in SIP it uses a Proxy Server instead of Gatekeeper and UDP as call setup protocol.
From the paper published in Columbia University with University of Wollongong, Australia and also from the paper completed by 3Com Corporation and Cisco Systems, it is find out that the time taken to setup a call using SIP between IP telephony and a PSTN is less than the time taken by H.323. In their experiment, they analyzed the call setup time of SIP signaling protocol on a real time network. They connected the calls between three various cities of USA at various networks with the help of proxy servers at the peak time. Then they measured the call setup time by measuring the time taken to reach each proxy servers between the end users. Then they find out the call setup time of SIP signaling protocol by calculating the time taken at each hops.
The SIP call setup delay is declared as 2~3 round trip time (one RTT approx to 90 to 100 milliseconds). It was also done on various scenarios like on various loads and various distances. [11] [12]

6 CONCLUSION

In our thesis, we experimentally analysed the call setup delay for H.323. Using the existing materials, we surveyed the call setup delay for SIP literally. We obtained an average call setup delay for H.323, at a peak time (heavy traffic) of almost 300~400 milliseconds and for SIP we studied this call setup delay as 200~300 milliseconds. The call setup time of H.323 signalling protocol is thus a little higher than the call setup time of H.323 signaling protocol. It is probably because of the Cisco devices we used for our experiments. The call setup time in traditional telephony is 4 minutes in 1923, it is reduced slowly and finally with the help of IP telephony the call setup time is reduced to 200~400 milliseconds in the current decade.

In our thesis we clearly studied and found that the call setup delay of SIP is a little less than H.323 but when we consider the other services of both the signaling protocols; we can say that H.323 is better than SIP. As the call setup performed in H.323 is done with the help of TCP, the connection is more reliable. However in SIP, the call setup is unreliable because of UDP. More-over the most important factor of H.323 is that, along with the audio packets it can transmit video and data packets with more reliability than SIP. Also, H.323 is more compatible; it can adopt any versions according to the users' necessity. When considering the bandwidth utilization, H.323 requires less bandwidth because it transmits the packets in binary format and in SIP it transfer in text format which requires more bandwidth than H.323. Even SIP has its own features like less call setup delay and flexibility as their advantages; the reliability, compatibility and bandwidth utilization of H.323 should also be taken into account as their advantages which are more essential.

Hence because of this, most of the organisations use H.323 even though they are expensive when compared to SIP.

7 References

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Appendix A:

Call Manager Configuration

```
Building configuration...
Current configuration : 2802 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname CME
!
boot-start-marker
boot-end-marker
!
!
no aaa new-model
memory-size iomem 10
dot11 syslog
!
!
ip cef
no ip dhcp use vrf connected
!
ip dhcp pool ITS
   network 192.58.7.0 255.255.255.0
   option 150 ip 192.58.7.1
   default-router 192.58.7.1
!
multilink bundle-name authenticated
!
voice-card 0
 no dspfarm
!
archive
 log config
  hidekeys
!
```

```
interface Loopback58
 ip address 192.58.7.1 255.255.255.0
!
interface FastEthernet0/0
 bandwidth 64
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface FastEthernet0/1
 ip address 192.168.50.2 255.255.255.0
 duplex auto
 speed auto
!
router eigrp 100
 network 10.0.0.0
 network 192.0.0.0 0.255.255.255
 no auto-summary
!
ip forward-protocol nd
!
 ip http server
no ip http secure-server
!
tftp-server flash:p
tftp-server flash:P00307020200.bin
tftp-server flash:P00307020200.loads
tftp-server flash:P00307020200.sb2
tftp-server flash:P00307020200.sbn
tftp-server flash:P00403020214.bin
!
control-plane
!
voice-port 0/2/0
 signal groundStart
 cptone SE
 impedance complex2
 music-threshold -70
 caller-id enable
!
```

```
voice-port 0/2/1
 signal groundStart
 cptone SE
 impedance complex2
 music-threshold -70
!
dial-peer voice 1 pots
 destination-pattern 5556001
 port 0/2/0
 forward-digits all
!
dial-peer voice 2 pots
 destination-pattern 5559
 port 0/2/1
 forward-digits all
!
gateway
 timer receive-rtp 1200
!
telephony-service
 load 7910 P00403020214
 load 7960-7940 P00307020200
 max-ephones 6
 max-dn 6
 ip source-address 192.58.7.1 port 2000
 auto assign 1 to 6
 dialplan-pattern 1 5... extension-length 4
 voicemail 5
 max-conferences 8 gain -6
 transfer-system full-consult
!
ephone-dn  1  dual-line
 number 5551
 call-forward busy 5
 call-forward noan 5 timeout 18
!
ephone-dn  2  dual-line
 number 5552
 call-forward busy 5
 call-forward noan 5 timeout 18
```

```
!
ephone-dn  3  dual-line
 number 5553
 call-forward busy 5
 call-forward noan 5 timeout 18
!
ephone-dn  4  dual-line
 number 5554
 call-forward busy 5
 call-forward noan 5 timeout 18
!
ephone-dn  5  dual-line
 number 5555
 call-forward busy 5
 call-forward noan 5 timeout 18
!
ephone-dn  6  dual-line
 number 5556
 call-forward busy 5
 call-forward noan 5 timeout 18
!
ephone  1
 no multicast-moh
 mac-address 0018.8B84.BE84
 type CIPC
 button  1:1
!
ephone  2
 no multicast-moh
 mac-address 0018.8B83.3CBC
 type CIPC
 button  1:2
!
ephone  3
 no multicast-moh
!
ephone  4
 no multicast-moh
!
ephone  5
 no multicast-moh
```

```
!
ephone  6
 no multicast-moh
!
line con 0
line aux 0
line vty 0 4
 login
!
scheduler allocate 20000 1000
!
end
```

IP Interface Brief:

Interface	IP-Address	OK?	Method	Status	Protocol
FastEthernet0/0	unassigned	YES	manual	administratively down	down
FastEthernet0/1	192.168.50.2	YES	manual	up	up
Loopback58	192.58.7.1	YES	manual	up	up

IP Routing:

```
Codes: C - connected, S - static, R - RIP, M - mobile, B - BGP
       D - EIGRP, EX - EIGRP external, O - OSPF, IA - OSPF inter area
       N1 - OSPF NSSA external type 1, N2 - OSPF NSSA external type 2
       E1 - OSPF external type 1, E2 - OSPF external type 2
       i - IS-IS, su - IS-IS summary, L1 - IS-IS level-1, L2 - IS-IS level-2
       ia - IS-IS inter area, * - candidate default, U - per-user static route
       o - ODR, P - periodic downloaded static route

Gateway of last resort is not set

D    192.168.14.0/24 [90/30720] via 192.168.50.1, 02:19:38, FastEthernet0/1
D    192.168.4.0/24 [90/35840] via 192.168.50.1, 01:14:01, FastEthernet0/1
     10.0.0.0/24 is subnetted, 2 subnets
D       10.10.1.0 [90/30720] via 192.168.50.1, 02:19:38, FastEthernet0/1
D       10.10.2.0 [90/33280] via 192.168.50.1, 02:19:38, FastEthernet0/1
C    192.58.7.0/24 is directly connected, Loopback58
```

D 192.168.7.0/24 [90/30720] via 192.168.50.1, 02:19:38, FastEthernet0/1
C 192.168.50.0/24 is directly connected, FastEthernet0/1
D 192.168.3.0/24 [90/30720] via 192.168.50.1, 02:19:39, FastEthernet0/1

Appendix B

Gateway Configuration

Building configuration...

Current configuration : 2127 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname GW
!
boot-start-marker
boot-end-marker
!
no aaa new-model
memory-size iomem 5
dot11 syslog
ip cef
!
no ip dhcp use vrf connected
!
ip dhcp pool its
 network 192.168.7.0 255.255.255.0
 option 150 ip 192.168.7.1
 default-router 192.168.7.1
!
multilink bundle-name authenticated
!
voice-card 0
 codec complexity high
!
archive
 log config

```
 hidekeys
!
interface FastEthernet0/0
 no ip address
 duplex auto
 speed auto
!
interface FastEthernet0/0.3
 encapsulation dot1Q 10
 ip address 192.168.3.1 255.255.255.0
!
interface FastEthernet0/0.7
 encapsulation dot1Q 20
 ip address 192.168.7.1 255.255.255.0
 h323-gateway voip interface
 h323-gateway voip id gk1.gw1.com ipaddr 10.10.2.1 1719
 h323-gateway voip h323-id gw1
!
interface FastEthernet0/0.13
 encapsulation dot1Q 13
 ip address 192.168.14.1 255.255.255.0
!
interface FastEthernet0/0.50
 encapsulation dot1Q 50
 ip address 192.168.50.1 255.255.255.0
!
interface FastEthernet0/1
 ip address 10.10.1.1 255.255.255.0
 duplex auto
 speed auto
!
interface Serial0/1/0
 no ip address
 shutdown
 no fair-queue
 clock rate 125000
!
interface Serial0/1/1
 no ip address
 shutdown
```

```
!
router eigrp 100
 network 10.0.0.0
 network 192.168.0.0 0.0.255.255
 no auto-summary
!
ip forward-protocol nd
!
 ip http server
no ip http secure-server
!
tftp-server flash:P00307020200.bin
tftp-server flash:P00307020200.loads
tftp-server flash:P00307020200.sb2
tftp-server flash:P00307020200.sbn
!
control-plane
!
dial-peer voice 1 pots
 destination-pattern 5556
 forward-digits all
!
dial-peer voice 3 pots
 destination-pattern 5556003
 forward-digits all
!
dial-peer voice 2 pots
 destination-pattern 5559
 forward-digits all
 no register e164
!
dial-peer voice 5 pots
 destination-pattern 5.....
 forward-digits all
!
dial-peer search type voice data
gateway
 timer receive-rtp 1200
!
line con 0
line aux 0
```

line vty 0 4
 login
!
scheduler allocate 20000 1000
end

IP Interface Brief

Interface	IP-Address	OK?	Method	Status	Protocol
FastEthernet0/0	unassigned	YES	manual	up	up
FastEthernet0/0.3	192.168.3.1	YES	manual	up	up
FastEthernet0/0.7	192.168.7.1	YES	manual	up	up
FastEthernet0/0.13	192.168.14.1	YES	manual	up	up
FastEthernet0/0.50	192.168.50.1	YES	manual	up	up
FastEthernet0/1	10.10.1.1	YES	manual	up	up
Serial0/1/0	unassigned	YES	unset	administratively down	down
Serial0/1/1	unassigned	YES	unset	administratively down	down

IP Routing

Codes: C - connected, S - static, R - RIP, M - mobile, B - BGP
D - EIGRP, EX - EIGRP external, O - OSPF, IA - OSPF inter area
N1 - OSPF NSSA external type 1, N2 - OSPF NSSA external type 2
E1 - OSPF external type 1, E2 - OSPF external type 2
i - IS-IS, su - IS-IS summary, L1 - IS-IS level-1, L2 - IS-IS level-2
ia - IS-IS inter area, * - candidate default, U - per-user static route
o - ODR, P - periodic downloaded static route

Gateway of last resort is not set

C 192.168.14.0/24 is directly connected, FastEthernet0/0.13
D 192.168.4.0/24 [90/33280] via 10.10.1.2, 01:16:27, FastEthernet0/1
10.0.0.0/24 is subnetted, 2 subnets
C 10.10.1.0 is directly connected, FastEthernet0/1
D 10.10.2.0 [90/30720] via 10.10.1.2, 02:38:23, FastEthernet0/1

D 192.58.7.0/24 [90/156160] via 192.168.50.2, 02:22:05, FastEthernet0/0.50
C 192.168.7.0/24 is directly connected, FastEthernet0/0.7
C 192.168.50.0/24 is directly connected, FastEthernet0/0.50
C 192.168.3.0/24 is directly connected, FastEthernet0/0.3

Appendix C

Gatekeeper Configuration

```
Building configuration...

Current configuration : 1727 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname GK
!
boot-start-marker
boot-end-marker
!
logging message-counter syslog
!
no aaa new-model
memory-size iomem 10
!
dot11 syslog
ip source-route
!
ip cef
!
ip dhcp pool GK-Client
   network 192.168.4.0 255.255.255.0
   default-router 192.168.4.1
!
no ipv6 cef
!
multilink bundle-name authenticated
!
voice-card 0
```

```
!
archive
 log config
  hidekeys
!
interface FastEthernet0/0
 ip address 192.168.4.1 255.255.255.0
 ip broadcast-address 192.168.4.0
 duplex auto
 speed auto
!
interface FastEthernet0/1
 description to ISPSw
 ip address 10.10.2.1 255.255.255.0
 ip broadcast-address 10.10.2.0
 duplex auto
 speed auto
!
interface Serial0/0/0
 no ip address
 ip broadcast-address 0.0.0.0
 shutdown
 no fair-queue
!
interface Serial0/0/1
 no ip address
 ip broadcast-address 0.0.0.0
 shutdown
 clock rate 125000
!
router eigrp 100
 network 10.0.0.0
 network 192.168.0.0 0.0.255.255
 no auto-summary
!
ip forward-protocol nd
ip route 10.1.1.0 255.255.255.0 10.1.1.6
ip route 10.1.1.0 255.255.255.0 10.1.6.1
ip route 10.1.1.0 255.255.255.0 10.1.6.6
ip http server
```

```
no ip http secure-server
!
nls resp-timeout 1
cpd cr-id 1
!
control-plane
!
ccm-manager fax protocol cisco
!
gatekeeper
 zone local gk1.gw1.com gw1.com 10.10.2.1
 zone local gk1.w1.com gw1.com
 zone remote gk2.gw2.com gw2.com 10.10.7.1 1719
 zone prefix gk1.gw1.com 17..
 zone prefix gk1.gw1.com 50..
 gw-type-prefix 1#* default-technology
 no shutdown
!
line con 0
line aux 0
line vty 0 4
 login
!
scheduler allocate 20000 1000
end
```

IP Routing

```
Codes: C - connected, S - static, R - RIP, M - mobile, B - BGP
       D - EIGRP, EX - EIGRP external, O - OSPF, IA - OSPF inter area
       N1 - OSPF NSSA external type 1, N2 - OSPF NSSA external type 2
       E1 - OSPF external type 1, E2 - OSPF external type 2
       i - IS-IS, su - IS-IS summary, L1 - IS-IS level-1, L2 - IS-IS level-2
       ia - IS-IS inter area, * - candidate default, U - per-user static route
       o - ODR, P - periodic downloaded static route

Gateway of last resort is not set

D    192.168.14.0/24 [90/33280] via 10.10.2.2, 02:41:07, FastEthernet0/1
C    192.168.4.0/24 is directly connected, FastEthernet0/0
     10.0.0.0/24 is subnetted, 2 subnets
```

D 10.10.1.0 [90/30720] via 10.10.2.2, 02:41:10, FastEthernet0/1
C 10.10.2.0 is directly connected, FastEthernet0/1
D 192.58.7.0/24 [90/161280] via 10.10.2.2, 02:24:52, FastEthernet0/1
D 192.168.7.0/24 [90/33280] via 10.10.2.2, 02:41:07, FastEthernet0/1
D 192.168.50.0/24 [90/33280] via 10.10.2.2, 02:41:07, FastEthernet0/1
D 192.168.3.0/24 [90/33280] via 10.10.2.2, 02:41:08, FastEthernet0/1

IP Interface Brief

Interface	IP-Address	OK?	Method	Status	Protocol
FastEthernet0/0	192.168.4.1	YES	manual	up	up
FastEthernet0/1	10.10.2.1	YES	manual	up	up
Serial0/0/0	unassigned	YES	manual	administratively down	down
Serial0/0/1	unassigned	YES	manual	administratively down	down

Appendix D

Internet Configuration

Building configuration...

Current configuration : 2483 bytes
!
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname INT
!
no aaa new-model
vtp domain TSHOOT
vtp mode transparent
ip subnet-zero
ip routing
!
no file verify auto

```
spanning-tree mode pvst
spanning-tree extend system-id
!
vlan internal allocation policy ascending
!
vlan 10
 name Data
!
vlan 20
 name Voice
!
vlan 30
 name GW1
!
vlan 40
 name GK1
!
vlan 70
 name Monitoring
!
interface FastEthernet0/1
 switchport access vlan 30
 switchport trunk encapsulation dot1q
 switchport mode access
!
interface FastEthernet0/2
 switchport access vlan 40
 switchport trunk encapsulation dot1q
 switchport mode access
!
interface FastEthernet0/3
 switchport access vlan 50
 switchport trunk encapsulation dot1q
 switchport mode access
!
interface FastEthernet0/4
 switchport access vlan 60
 switchport trunk encapsulation dot1q
 switchport mode access
!
interface FastEthernet0/5
```

```
 no switchport
 no ip address
!
interface FastEthernet0/6
 switchport mode access
!
interface FastEthernet0/7
 no switchport
 ip address 10.10.1.2 255.255.255.0
!
interface FastEthernet0/8
 no switchport
 ip address 10.10.2.2 255.255.255.0
!
interface FastEthernet0/9
 switchport mode access
!
interface FastEthernet0/10
 switchport mode access
!
interface FastEthernet0/11
 switchport mode access
!
interface FastEthernet0/12
 switchport mode access
!
interface FastEthernet0/13
 switchport access vlan 70
 switchport mode access
!
interface FastEthernet0/14
!
interface FastEthernet0/15
!
interface FastEthernet0/16
!
interface FastEthernet0/17
!
interface FastEthernet0/18
!
```

```
interface FastEthernet0/19
!
interface FastEthernet0/20
!
interface FastEthernet0/21
!
interface FastEthernet0/22
!
interface FastEthernet0/23
!
interface FastEthernet0/24
!
interface GigabitEthernet0/1
!
interface GigabitEthernet0/2
!
interface Vlan1
 no ip address
 shutdown
!
ip classless
ip http server
ip http secure-server
!
control-plane
!
line con 0
 logging synchronous
line vty 0 4
 login
line vty 5 15
 login
!
monitor session 1 source vlan 10 - 60 rx
monitor session 1 destination interface Fa0/13
end
```

IP Interface Brief

Interface	IP-Address	OK?	Method	Status	Protocol
Vlan1	unassigned	YES	manual	administratively down	down
FastEthernet0/1	unassigned	YES	unset	down	down

FastEthernet0/2	unassigned	YES unset	down	down
FastEthernet0/3	unassigned	YES unset	down	down
FastEthernet0/4	unassigned	YES unset	down	down
FastEthernet0/5	unassigned	YES manual	down	down
FastEthernet0/6	unassigned	YES unset	down	down
FastEthernet0/7	10.10.1.2	YES manual	up	up
FastEthernet0/8	10.10.2.2	YES manual	up	up
FastEthernet0/9	unassigned	YES unset	down	down
FastEthernet0/10	unassigned	YES unset	down	down
FastEthernet0/11	unassigned	YES unset	down	down
FastEthernet0/12	unassigned	YES unset	down	down
FastEthernet0/13	unassigned	YES unset	down	down
FastEthernet0/14	10.10.70.1	YES manual	down	down
FastEthernet0/15	unassigned	YES unset	down	down
FastEthernet0/16	unassigned	YES unset	down	down
FastEthernet0/17	unassigned	YES unset	down	down
FastEthernet0/18	unassigned	YES manual	administratively down	down
FastEthernet0/19	unassigned	YES unset	down	down
FastEthernet0/20	unassigned	YES unset	down	down
FastEthernet0/21	unassigned	YES unset	down	down
FastEthernet0/22	unassigned	YES unset	down	down
FastEthernet0/23	unassigned	YES unset	down	down
FastEthernet0/24	unassigned	YES unset	down	down
GigabitEthernet0/1	unassigned	YES unset	do	down
GigabitEthernet0/2	unassigned	YES unset	down	down

IP Routing

Codes: C - connected, S - static, R - RIP, M - mobile, B - BGP
D - EIGRP, EX - EIGRP external, O - OSPF, IA - OSPF inter area
N1 - OSPF NSSA external type 1, N2 - OSPF NSSA external type 2
E1 - OSPF external type 1, E2 - OSPF external type 2, E - EGP
i - IS-IS, su - IS-IS summary, L1 - IS-IS level-1, L2 - IS-IS level-2
ia - IS-IS inter area, * - candidate default, U - per-user static route
o - ODR, P - periodic downloaded static route

Gateway of last resort is not set

D 192.168.14.0/24 [90/30720] via 10.10.1.1, 02:47:52, FastEthernet0/7
D 192.168.4.0/24 [90/30720] via 10.10.2.1, 01:25:59, FastEthernet0/8
10.0.0.0/24 is subnetted, 2 subnets

```
C       10.10.1.0 is directly connected, FastEthernet0/7
C       10.10.2.0 is directly connected, FastEthernet0/8
D    192.58.7.0/24 [90/158720] via 10.10.1.1, 02:31:36, FastEthernet0/7
D    192.168.7.0/24 [90/30720] via 10.10.1.1, 02:47:52, FastEthernet0/7
D    192.168.50.0/24 [90/30720] via 10.10.1.1, 02:47:52, FastEthernet0/7
D    192.168.3.0/24 [90/30720] via 10.10.1.1, 02:47:52, FastEthernet0/7
```

Appendix E

Traffic Generator Configuration

```
Building configuration...

Current configuration : 982 bytes
!
version 12.4
service timestamps debug datetime msec
service timestamps log datetime msec
no service password-encryption
!
hostname Router
!
boot-start-marker
boot-end-marker
!
no aaa new-model
memory-size iomem 5
!
ip cef
!
ip host PAGENT-SECURITY-V3 97.32.43.85 87.84.0.0
!
multilink bundle-name authenticated
!
!
voice-card 0
 no dspfarm
!
interface FastEthernet0/0
 bandwidth 32
 ip address 192.168.3.10 255.255.255.0
 duplex auto
```

```
 speed auto
!
interface FastEthernet0/1
 no ip address
 shutdown
 duplex auto
 speed auto
!
interface Serial0/0/0
 ip address 172.16.34.4 255.255.255.248
!
interface Serial0/0/1
 no ip address
 shutdown
 clock rate 2000000
!
router eigrp 100
 network 192.0.0.0 0.255.255.255
 no auto-summary
!
ip http server
no ip http secure-server
!
control-plane
!
line con 0
 exec-timeout 0 0
line aux 0
line vty 0 4
 login
!
scheduler allocate 20000 1000
!
end
```

IP Routing

```
Codes: C - connected, S - static, R - RIP, M - mobile, B - BGP
       D - EIGRP, EX - EIGRP external, O - OSPF, IA - OSPF inter area
       N1 - OSPF NSSA external type 1, N2 - OSPF NSSA external type 2
```

E1 - OSPF external type 1, E2 - OSPF external type 2
i - IS-IS, su - IS-IS summary, L1 - IS-IS level-1, L2 - IS-IS level-2
ia - IS-IS inter area, * - candidate default, U - per-user static route
o - ODR, P - periodic downloaded static route

Gateway of last resort is not set

D 192.168.14.0/24 [90/80005120] via 192.168.3.1, 01:52:55, FastEthernet0/0
D 192.168.4.0/24 [90/80010240] via 192.168.3.1, 01:16:11, FastEthernet0/0
10.0.0.0/24 is subnetted, 2 subnets
D 10.10.1.0 [90/80005120] via 192.168.3.1, 01:52:55, FastEthernet0/0
D 10.10.2.0 [90/80007680] via 192.168.3.1, 01:52:55, FastEthernet0/0
D 192.58.7.0/24 [90/80133120] via 192.168.3.1, 01:52:55, FastEthernet0/0
D 192.168.7.0/24 [90/80005120] via 192.168.3.1, 01:52:55, FastEthernet0/0
D 192.168.50.0/24 [90/80005120] via 192.168.3.1, 01:52:57, FastEthernet0/0
C 192.168.3.0/24 is directly connected, FastEthernet0/0

IP Interface Brief

Interface	IP-Address	OK?	Method	Status	Protocol
FastEthernet0/0	192.168.3.10	YES	manual	up	up
FastEthernet0/1	unassigned	YES	manual	administratively down	down
Serial0/0/0	172.16.34.4	YES	NVRAM	down	down
Serial0/0/1	unassigned	YES	manual	administratively down	down

Appendix F

Switch Configuration

Building configuration...

Current configuration : 1930 bytes
!
version 12.2
no service pad
service timestamps debug uptime
service timestamps log uptime
no service password-encryption
!
hostname Switch

```
!
no aaa new-model
system mtu routing 1500
ip subnet-zero
!
no file verify auto
spanning-tree mode pvst
spanning-tree extend system-id
!
vlan internal allocation policy ascending
!
interface FastEthernet0/1
 switchport trunk encapsulation dot1q
 switchport mode trunk
!
interface FastEthernet0/2
!
interface FastEthernet0/3
 switchport access vlan 10
 switchport mode access
!
interface FastEthernet0/4
 switchport access vlan 10
 switchport mode access
!
interface FastEthernet0/5
 switchport access vlan 10
 switchport mode access
!
interface FastEthernet0/6
 switchport access vlan 10
 switchport mode access
!
interface FastEthernet0/7
 switchport access vlan 20
 switchport mode access
!
interface FastEthernet0/8
 switchport mode access
!
```

```
interface FastEthernet0/9
 switchport access vlan 80
 switchport mode access
!
interface FastEthernet0/10
 switchport mode access
!
interface FastEthernet0/11
!
interface FastEthernet0/12
!
interface FastEthernet0/13
 switchport access vlan 13
 switchport mode access
!
interface FastEthernet0/14
!
interface FastEthernet0/15
!
interface FastEthernet0/16
!
interface FastEthernet0/17
!
interface FastEthernet0/18
!
interface FastEthernet0/19
!
interface FastEthernet0/20
!
interface FastEthernet0/21
!
interface FastEthernet0/22
!
interface FastEthernet0/23
!
interface FastEthernet0/24
 switchport access vlan 50
 switchport mode access
!
interface GigabitEthernet0/1
!
```

```
interface GigabitEthernet0/2
!
interface Vlan1
 no ip address
 shutdown
!
interface Vlan10
 no ip address
!
interface Vlan13
 ip address 192.168.14.2 255.255.255.0
!
ip classless
ip http server
!
control-plane
!
line con 0
line vty 0 4
 login
line vty 5 15
 login
!
monitor session 1 source vlan 10 - 20 rx
monitor session 1 destination interface Fa0/13
end
```

Vlan

```
VLAN Name                        Status    Ports
---- ------------------------------ --------- -------------------------------
1    default                      active    Fa0/2, Fa0/8, Fa0/10, Fa0/11
                                            Fa0/12, Fa0/14, Fa0/15, Fa0/16
                                            Fa0/17, Fa0/18, Fa0/19, Fa0/20
                                            Fa0/21, Fa0/22, Fa0/23, Gi0/1
                                            Gi0/2
10   data                         active    Fa0/3, Fa0/4, Fa0/5, Fa0/6
13   VLAN0013                     active    Fa0/13
15   VLAN0015                     active
20   voice                        active    Fa0/7
```

```
50   PBX                              active    Fa0/24
80   traffic                          active    Fa0/9
1002 fddi-default                     act/unsup
1003 token-ring-default               act/unsup
1004 fddinet-default                  act/unsup
1005 trnet-default                    act/unsup

VLAN Type  SAID       MTU   Parent RingNo BridgeNo Stp  BrdgMode Trans1 Trans2
---- ----- ---------- ----- ------ ------ -------- ---- -------- ------ ------
1    enet  100001     1500  -      -      -        -    -        0      0
10   enet  100010     1500  -      -      -        -    -        0      0

VLAN Type  SAID       MTU   Parent RingNo BridgeNo Stp  BrdgMode Trans1 Trans2
---- ----- ---------- ----- ------ ------ -------- ---- -------- ------ ------
13   enet  100013     1500  -      -      -        -    -        0      0
15   enet  100015     1500  -      -      -        -    -        0      0
20   enet  100020     1500  -      -      -        -    -        0      0
50   enet  100050     1500  -      -      -        -    -        0      0
80   enet  100080     1500  -      -      -        -    -        0      0
1002 fddi  101002     1500  -      -      -        -    -        0      0
1003 tr    101003     1500  -      -      -        -    -        0      0
1004 fdnet 101004     1500  -      -      -        ieee -        0      0
1005 trnet 101005     1500  -      -      -        ibm  -        0      0

Remote SPAN VLANs
------------------------------------------------------------------------------


Primary Secondary Type              Ports
------- --------- ----------------- ------------------------------------------
```